NEAR EAST UNIVERSITY

Faculty of Engineering

Department of Electrical and Electronic Engineering

TELECOMMUNICATIONS NETWORK

Graduation Project
EE- 400

Student: Matiyya Bannoura (970714)

Supervisor: Professor Fakhreddin Mamedov

Lefkosa - 2001

TABLE OF CONTENTS

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TABLE OF CONTENTS

ACKNOWLEDGMENT

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I am and shall always be in your gratitude. With all my love, respect and best wishes, thank you all so much.

ABBREVIATIONS

PTT	Post, Telegraph and Telephone.
PSTN	Public witching Telephone Network.
PLMN	Public Land Mobile Network.
MAN	Metropolitan Area Network.
WAN	Wide Area Network.
LAN	Local Area Network.
ATM	Asynchronous Transfer Mode.
ISDN	Integral Service Digital Network.
SONET	Synchronous Optical Network.
SMDS	Switched Multi-megabyte Data Service.
FDDI	Fiber Distributed Data Network.
CO	Central Office.
AT&T	American Telephone & Telegraph.
RBOC	Regional Bell Operating Company.
NCR	National Cash Register.
GTE	General Telephone and Electric.
CATV	Common Antenna Television.
PCS	Personal Communications Services.
MFS	Macintosh File System.
MCI	Media Control Interface.
FCC	Federal Communication Commission.
IXC	Inter Exchange Carrier.
SNET	Southern New England Telephone.
TCI	Tele-Communications Inc.
CLEC	Competitive Local Exchange Carrier.
DSL	Digital Subscriber Line.
TELCOS	Name of a Telephone Company.
OSS	Operation Support System.
COE	Central Office Equipment.
DMS	Digital Multiplex System.
FMT	Fiber Multiplex Transport.
DACS	Digital Access and Cross-Connect System.
RCU	Remote Carrier Urban.
RSC	Remote Switch Concentrator.
RLCM	Remote Line Concentrating Module.
FMS	Forms Management System.
MFA	Mechanized Frame Administration.
IP	Internet Protocol.
SLAT	System Line Up and Test.
WDM	Wavelength Division Multiplexing.
PON	Passive Optical Network.
GSM	Global System Mobile.
CDMA	Code Division Multiple Access.
N-CDMA	Narrowband-CDMA.

UMTS	Universal Mobile Telecommunication service.
TDMA	Time Division Multiple Access.
DECT	Digital European Cordless Telecommunications.
PAC	Privilege Attribute Certificate.
SDH	Synchronous Digital Hierarchy.
WLL	Wireless Local Loop.
RLL	Radio Local Loop.
VoD	Video on Demand.
VPN	Virtual Private Network.
CTI	Computer Telephone Integration.
VOIP	Voice Over Internet Protocols.
CBR	Constant Bit Rate.
PCM	Pulse Code Modulation.
B-ISDN	Broadband-ISDN.
HDTV	High Definition Television.
DPCM	Deferential PCM.
ASCII	American Standard Code for Information Interchange.
ASK	Amplitude Shift Keying.
FSK	Frequency Shift Keying.
PSK	Phase Shift Keying.
DTE	Data Terminal Equipment.
DM	Delta Modulation.
FM	Frequency Modulation.
AM	Amplitude Modulation.
PM	Phase Modulation.
PAM	Phase Amplitude Modulation.
QAM	Quadrature Amplitude Modulation.
BER	Bit Error Rate.
MNP	Microcosm's Networking Protocol.
LAPM	Link Access Procedure for Modems.
CRC	Cyclic Redundancy Checking.
ARQ	Automatic Repeat Request.
FEC	Forward Error Correction.
TCM	Trellis Code Modulation.
PDU	Protocol Data Units.
OSIRM	Open System Interconnection Reference Module.
SDLC	Synchronous Data Link Control.
HDLC	High Level Data Link Control.
LAP-B	Link Access Procedure B.
PUC	Public Utility Commission.
SCC	Specialized Common Carrier.
VAN	Value Added Network.
BOC	Bell Operating Company.
ACD	Automatic Call Distributor.
ANI	Automatic Data Identification.
CPE	Customer Premises Equipment.
S/N	Signal to Noise Ratio.

CD	Compact Disk.
CCITT	Consultive Committee on International Telephone and Telegraph.
ADPCM	Adaptive Pulse Code Modulation.
CVSD	Continuously Variable Slope Delta.
FDM	Frequency Division Multiplexing.
SSB	Signal Side Band.
LEC	Local Exchange Carrier.
CSA	Carrier Serving Area.
ESS	Electronic Switching System.
CRT	Cathode Ray Tubes.
DTE	Data Terminal Equipment.
DCE	Data Circuit-Terminating Equipment.
EBCDIC	Extended Binary Coded Decimal Interchange Code.
LATA	Local Access and Transport Area.

ABSTRACT

Telecommunications is one of the fastest growing business sectors of modern information technologies. Today telecommunications include a vast variety of modern technologies and services. Some services, such as the fixed telephone service in developed countries are becoming mature; and some are exploding like cellular mobile communications. The new environment provides new options for users and we should be more aware of telecommunications as a whole, these services and options are explained in the project, besides that special attention I paid to the security aspects and cost of telecommunications network.

We will discuss the basic purpose of telecommunications network and see that telecommunications network consists of many different networks providing services for the users. Also the three technologies needed for communication through a network are discussed briefly, besides the three categories that networks consist of.

In general this project aims to give a basic understanding of the structure and operation of a telecommunications network. A deeper theory of telecommunications, such as the spectral analysis or signals or detailed knowledge of the operation and functions of data and voice networks.

INTRODUCTION

Telecommunications network today has become one of the most important technologies in the human beings life, since it reduces distances, cost, time and has the ability to provide information from any region connected to this network, it is also established fact that telecommunications serve as a tool for development for the whole world.

Every life is dependent on telecommunications. Each of us uses telecommunications services and services that rely on telecommunications daily, such as banking; automatic teller machines, telebanking, aviation; booking of tickets, booking hotel rooms by travel agencies and a lot of services.

This thesis aims to give answers to the fundamental questions concerning telecommunications network and services, telecommunications as a business area and the general trends of technical development. Some of these questions are: what is the structure and what are the main components of a modern telecommunications network? And what are the future developments of telecommunications network? Starting from the structure and components and ending with the future developments.

The thesis consists of introduction, three chapters and conclusion.

Chapter 1 introduces telecommunications network, an overview, historical view, the future and understanding the process of telecommunications network.

Chapter 2 is studying telecommunications network in details, the components of telecommunications network, telecommunication carriers and design philosophies.

Chapter 3 discus's the future developments of telecommunications network.

Conclusion presents the significant results, contribution and future investigation.

CHAPTER 1

INTRODUCTION TO TELECOMMUNICATIONS

1.1 Introductions and Overview

We are now in what is called the "Information Age." Information has become a commodity not only to the business community, but also to all of society. In fact, the whole economic well being of a nation may well depend on the telecommunication infrastructure in place in that nation, and the reach of that infrastructure to other nations. Just as there are techniques on how best to handle physical commodities, so there are techniques on how best to manage information, and how to transmit it in a timely fashion to where it is needed.

Everyone of us hears the word telecommunications many times everyday, but less of people who knows the meaning of this word, so let us ask, what is telecommunications?

The word communication derives from the Latin word communicare: to impart, participate. The term "information" can be viewed as a primitive (axiomatic) concept, requiring no further definition. The science of "communication" is the study of all information transfer processes.

Telecommunications has been defined as a technology concerned with communicating from a distance, and we can categorize it in various ways. Figure 1.1 shows one possible division. It includes mechanical communication and electrical communication because telecommunications has developed from mechanical to electrical using increasingly more sophisticated electrical systems. This is why many authorities such as national Post, Telegraph, and Telephone (PTTs) are involved in telecommunications by both means.

Our main concern here is electrical and bi-directional communication, as shown in the upper part of Figure 1.1. The share of the mechanical telecommunications such as conventional mail and press is expected to decrease; while the electrical, especially bi-directional, communication will increase and take the major share of overall turnover of telecommunications in the future. Hence, major press corporations are interested in electrical telecommunications as a business opportunity. Telecommunication is expected to be one of the most rapidly growing business sectors during the next few years.

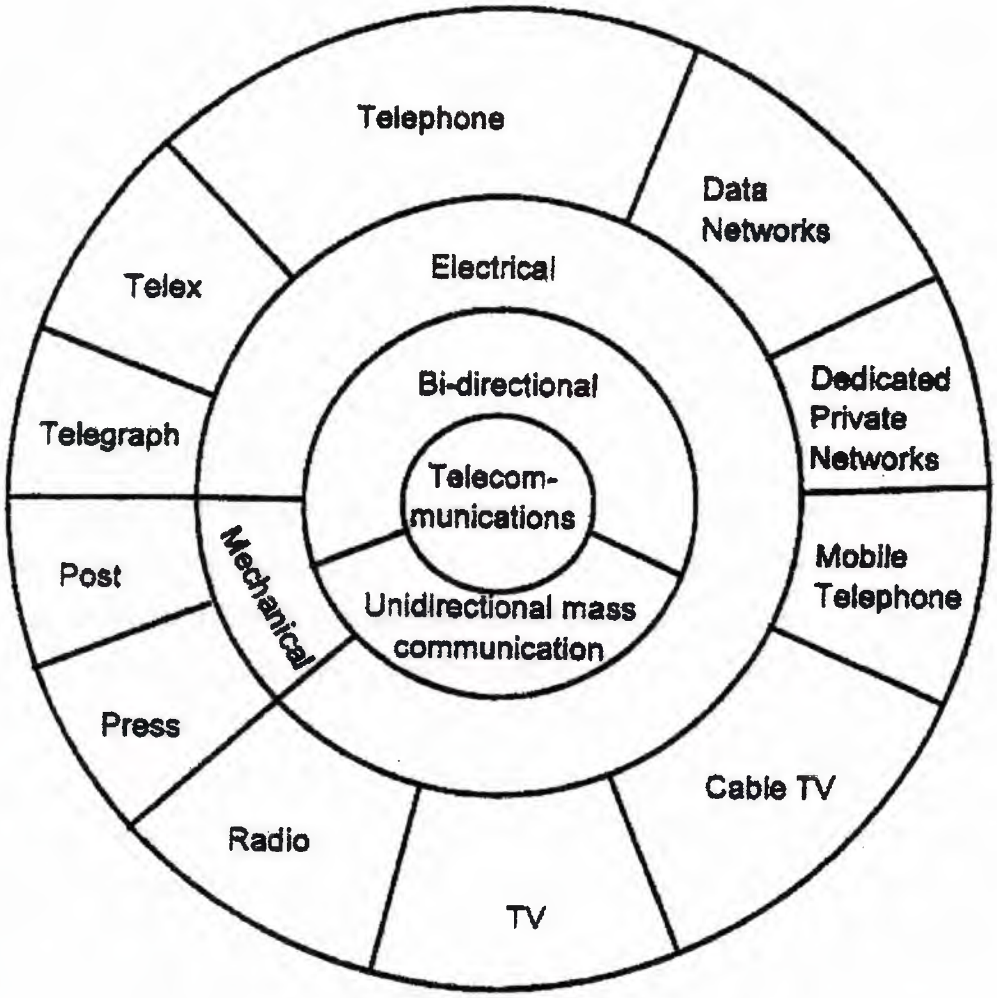


Figure 1.1 Telecommunications.

Given two entities A and B, communication is the action of transferring information from A to B, and vice versa. A and B are communicating when A is suitably able to code a message of information and relay it to B through an appropriate medium, while B is suitably equipped to receive the message by using or interpreting it in some fashion. A set of entities A, B, C, . . ., is in mutual communication when the pairs (A,B), (A,C), (B.C), etc. can ,at appropriate times, communicate with other. Protocols are agreed conventions between communicating entities on how to carry out mechanics of the communication process.

Telecommunication entails disciplines, means, and methodologies to communicate over distances, in effect, to transmit voice, video, facsimile, and computer data. Data communication entails disciplines, means, and methodologies particular to transmission of computer data, possibly over a specially engineered network, and typically from a protocol perspective. The data communication field is a subset of the telecommunication field.

A telecommunications network is the combination of numerous network elements that are required to support voice, data, or video services in local or long-distance applications. A telecommunications network is the foundation of all telephony activity; it is the network that connects the end user to virtually anywhere in the world through the use of copper cable, coaxial cable, and fiber cable or through wireless technology such as microwave or satellite. Telecommunication provides communications over a distance using technology to overcome that distance. It usually means the transmission of words, sounds, pictures, or data in the form of electronic signals or impulses, sent either as an individual message between two parties or as a broadcast to be received at many locations. While broadcasting is far removed from private communications, a new range of one-to-one communication services (including video-on-demand, and other personal information and entertainment services provided over cable networks and so-called "web casting" over the Internet) will blur the current clear distinction between the two.

Telecommunications network is a system and method for controlling on a worldwide basis two or more telecommunications networks, which are they selves capable of exercising a form of common channel signaling network control. The system uses an architecture in which a destination telecommunications network having common channel signaling control is connected to an originating telecommunications network having common signaling control through a call set up and control methodology which provides ad hoc connection between the two spaced telecommunication networks and common channel signaling networks via an unrelated world wide data network which preferably constitutes the internet.

Since its invention by Alexander Graham Bell in 1876, the telephone has become the most familiar form of telecommunications. More recently, voice telephony has been supplemented by a range of computer-based telecommunication services. These have become popular through the Internet and World Wide Web-vast computer networks that provide many people with the means to exchange information.

Telecommunication system involves Public Switching Telephone Network (PSTN), Public Land Mobile Telephone Network (PLMN), radio and television network and the emergence worldwide computer. Technological advancements develop an exponential fashion in the telecommunications industry. Interconnection Local Area Network (LAN), Metropolitan Area Network (MAN) and Wide Area Network (WAN) via PSTN has created an advanced generation of switching technologies such as Asynchronous Transfer Mode (ATM), Integral Service Digital Network (ISDN), Synchronous Optical Network (SONET), Switched Multi-megabyte Data Service (SMDS), Fiber Distributed Data Network (FDDI).

The theme "Telecommunications and the Environment" is a particularly important subject, which could hardly be more relevant to the world today.

An outsider, and even someone working in an area specifically concerned with the environment, might well ask what telecommunications and the environment have to do with each other.

At first sight, the environment, in the broadest sense of the word, would seem to be unrelated to telecommunications. And yet, there are very real links between the two. They are in fact more or less inseparable.

It is a generally established fact that telecommunications serve as a tool for development. What kind of development do we mean? The answer is clear: sustainable development. If we first consider the problem of the environment and then define the concept of sustainable development, it should be easy to demonstrate the almost organic links between telecommunications and the environment, taking a look at existing technologies, the information they convey in all its forms and the extent to which they can provide solutions to the problem of environmental protection.

The deployment of the telecommunications network is the final stage of the process, and it requires experts from a number of different disciplines, including design (outside plant and central office [CO], construction (outside-plant cable placing), and CO equipment installation and testing. The most effective way to manage this stage of the process is to use a company that serves as a single point of contact with project-management expertise and that can manage every aspect of the entire job (see Figure 1.2).

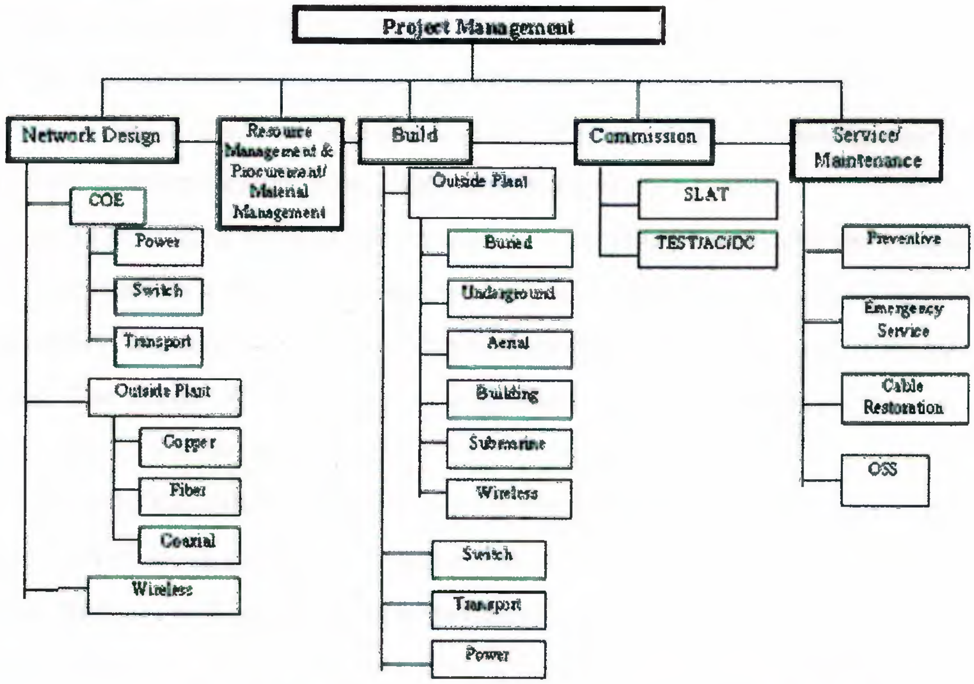


Figure 1.2 Deployment of a Telecommunications Network.

1.2 Historical View

It is now taken for granted in developed nations that by pressing a few buttons people can talk to family, friends, or business associates across the world. The technology that has led to one of the most complex creations of the 20th century, the telephone network has evolved over the past hundred years or so.

The first electrical means of communication was not the telephone, however, but the telegraph, which allowed messages, sent in code (usually Morse Code) to be received and printed at a distant location. The age of commercial telegraphy dawned in 1839 when the British pioneers William Fothergill Cooke and Charles Wheatstone opened their line alongside the main railway route running west from London. A technically simpler system of telegraphy was devised in 1843 by Samuel Morse, and after this the spread of telegraph networks was rapid, with routes spreading across most of the countries of the Old and New Worlds and then beneath the oceans that separated them. By 1930 nearly

650,000 km (400,000 mi) of undersea cables had been laid, linking the economic, political, military, and cultural institutions of the world.

An even greater breakthrough was made in 1876, when Alexander Graham Bell made the first telephone call to his assistant with the words "Mr. Watson, come here, I want you". Bell's invention sparked a series of innovations, ultimately culminating in today's information superhighway. Key steps along the way were:

In 1889 Almon Strowger developed an automatic switching system that could set up a telephone call without intervention by a human operator. Strowger's motivation for this invention was to prevent his calls being diverted to a business competitor by his local operator. The impact of the invention was much wider as it provided the basis for the current telephone network.

In 1901 Guglielmo Marconi demonstrated that radio waves could be used to transmit information over long distances when he sent a radio message across the Atlantic. Radio is still one of the key transmission media today, and is the basis of many mobile services.

In 1947 William Shockley, John Bardeen, and Walter Brattain invented the transistor. This enabled the electronics revolution to take place and provided the basis for a computerized, rather than mechanical, telecommunications network.

In 1965 Charles Kao put forward the theory that information could be carried using optical fibers. These have subsequently been developed to provide a means of carrying huge amounts of information at very high speed. Optical fibers form the backbone of the global transmission network.

1.2.1 The 1990s (The Century Closes - Into the Next Millennium)

1990- IBM sells ROLM Corporation to Germany based telecommunications giant - Siemens Corporation. AT&T develops the optical digital processor. Telmex (Mexico) is privatized. Telecom New Zealand is privatized.

1991-Bell Labs develop photonic switching. The Court of Appeals orders Judge Green to lift the ban on RBOC entry into information services. AT&T fights the long fight and buys NCR. GTE purchases Contel.

1992-The first color motion videophone introduced in the United States. The U.S. reaches its 10 millionth cellular subscriber. The Cable TV Act is introduced to regulate CATV pricing.

1993-The first digital mobile network is established in the U.S. (Los Angeles) while the first all digital cellular networks is brought up in Orlando, FL. The FCC allocates spectrum for PCS. Europe sets 1998 as the date for full liberalization of its telecom markets.

1994-The FCC begins PCS auctions. AT&T purchase McCaw Cellular. The number of subscriber telephones lines in the United States reaches 157.9 million; in the world: 609 million.

1995-There are now 25 million cellular subscribers in the U.S. Worldwide; 30 million users are now on the Internet.

1996-Commercial PCS operations begin in the U.S. The Cable Modem is introduced as the number of U.S. cellular subscribers reaches 40 million. AT&T announces its second major divestiture by spinning off NCR and its equipment business (including Bell Labs) under the Lucent Technologies name. Deutsche Telekom (Germany) is privatized. WorldCom and MFS merge to join local and long distance service providers. MCI and British Telecom merger to create Concert. France Telecom and Deutsche Telekom buy 10% of Sprint to form Global One alliance. Southwestern Bell announces merger plans with Pacific Telesis. Bell Atlantic announces plans to merge with NYNEX. The Telecommunications Act is the first real revision of the Communications Act of 1934 and is aimed at creating full competition in all markets.

1997-AT&T completes divestiture of Lucent Technologies and NCR. Implementation of the Telecommunication Act of 1996 is held up in some quarters in appeals courts as the RBOCs and the long distance companies' battle over requirements. Bell Atlantic wins approval of its takeover of NYNEX. SBC wins approval of its purchase of Pacific Telesis. The number of RBOCs now sits at five. Ameritech and BellSouth file, under the Telecom Act of 1996, to provide long distance services within their service areas. Both companies are refused permission to do so based on the lack of competition in their local markets. The RBOCs contest (and win) the FCCs authority on overseeing the opening of local telephone markets under the Telecommunications Act of 1996. British Telecom loses its bid for MCI to WorldCom. Lucent Technologies acquires Octel Communications for $1.8 billion. SBC wins a temporary victory when a Texas court rules that the Telecom Act of 1996 is anti-competitive by requiring the RBOCs to complete a series of steps to open their local markets while placing no such requirements on competitors (IXCs) wishing to enter the local markets.

1998-The FCC condemns the Texas Court's ruling and requests a decision from the Federal Courts. AT&T announces plans to acquire Teleport Communications. SBC announces its plan to acquire Southern New England Telephone (SNET) of Connecticut - in the heart of Bell Atlantic territory. AT&T announces plans to merge with TCI. WorldCom sells its Internet unit to Cable & Wireless. The WorldCom purchase of MCI is approved. SBC announces its plan to "merge" with Ameritech. Bell Atlantic announces plans to merge with GTE. Ameritech attempts to enter the long distance market using Qwest. Bell Atlantic and Bell South's bids to enter the long distance market are denied. The FCC's attempt to lower access fees and implement"universal service charges" results in higher costs to the end user and a stampede of user complaints.

1999-Organizations all over the world spend billions of dollars as they try to make their telecommunications systems and networks ready for the turn of the century. MCI WorldCom becomes official. Mergia Mania strikes the telecommunications industry as thousands of small start-up companies are purchased by larger ones. SBC and Ameritech announce plans to merge. The Internet envelopes the business community as companies scramble to ensure that they are ready to do business via this "World Wide Web". Bell Atlantic becomes the first "Baby Bell" to be approved to offer inter-LATA long distance services to customers in New York. The antiquated manner of telephone number distribution and the soaring number of competitive local exchange carriers (CLECs) places demands for new area codes throughout the country. The FCC allows the "universal service" percentage rate charged to IXC (and thus customers) to rise. Qwest and Global Crossing plans to merge fall through as Qwest moves to purchase "Baby Bell" US West. MCI WorldCom announces plans to merge with Sprint. The Federal Government continues its anti-trust suit against Microsoft Corporation.

1.2.2 The 2000s (The 21st Century Begins)

2000- Years of preparation and billions of dollars result in the Y2K "Bug" being nothing more than a minor pest on January 1st. Mergers run rampart in the telecommunications industry as companies plan for the future and the Internet. Lucent Technologies announces it will "spin off" its enterprise solution group into a new company. Bell Atlantic and GTE announce that their new combined company will be named "Verizon" (derived from the Latin for truth) thus moving the "Bell" name into its history. Cisco and Nortel networks jockey for position as leaders in voice and data communications and the Internet. Qwest divests itself of long distance companies in the US West service area in order to complete its purchase of US West. President Clinton requests that the "universal service" fee be increased to allow accommodation of Native American reservations and their technology needs. AT&T moves toward completion of its acquisition of Media One Communications. DSL becomes the new "hot service" for homes and small businesses accessing the Internet. The Federal Government rules against Microsoft Corporation - calls ring out for the organization to be divested into two separate companies. The Federal Trade Commission recommends rejection of the MCI WorldCom and Sprint merger putting this mega-deal at jeopardy. The Supreme Court upholds the FCC "detariffing" section of the Telecom Act of 1996 - All non-dominant carriers must conduct this detariffing over the next seven months (May - December). Lucent Technologies divest itself of its enterprise systems division creating Avaya Communications. WorldCom announces itself intent to buy Intermedia. British Telecom and AT&T officially call off rumored merger talks as AT&T hits financial hard times. First AT&T and then MCI announce major "spin offs" of primarily their consumer long distance divisions. Verizon wins the recommendation of Massachusetts's regulators to offer long distance service in Massachusetts; approval by the FCC is expected in December. The year turns into a financial disaster for many so called "dot com" companies as venture capital and high stock prices dry up resulting in layoffs, bankruptcies and the nickname "dot bombs". Verizon withdraws its application to provide long distance service to Massachusetts residents and companies. DSL providers nationwide fall on hard times as provisioning and maintaining the service becomes too expensive resulting in companies like Harvard Net and Digital Broadband to discontinue DSL service. SBC pays a record

$6.1 million fine for not meeting performance standards set as a condition of its merger with Ameritech.

The modern telephone network can be viewed as a globally distributed machine that operates as a single resource. Much of it uses interconnected computers. The network that most people use to carry voice traffic can also be used to transfer data in the form of pictures, text, and video images.

1.3 The Future

No one can predict with certainty exactly how telecommunications will develop, but certain trends can be noted. The cost of communication is falling in real terms, making advanced applications more affordable. Broader competition in the marketplace will reduce prices further. Telephone companies (telcos) recognize their revenues from carrying calls will decline and are encouraging, successfully, many new valued-added services that combine communication with the supply of information or services. Most of these have yet to evolve, but electronic commerce, mobile commerce, and various information-on-demand services are already being developed.

Some communication services currently provided by wire are migrating to radio means for greater convenience and flexibility; this includes not just cordless telephones in the home and the workplace but also connecting these telephones to the network.

Conversely radio and television programs, traditionally broadcast over the airwaves, are moving on to cable networks.

Fixed-mobile convergence is another trend, in which the distinction between conventional telephones and mobile networks will dwindle. Many people will carry a single "personal communicator" that functions as a cordless phone within the home, as their business extension in the workplace, and as a pocket mobile telephone elsewhere.

Today, thousands of information channels can be carried by a single underlying medium, like the fiber optic cables, which have a high bandwidth and a high transmission rate. A fiber optic link can carry multiple gigabit data rates. In the 10^{10} b/s range. An empirical review of the bandwidth that can be carried over transmission systems, shows that the bandwidth has increased by an order of magnitude every 20 years: 1950s: 10^{8} b/s; 1970s: 10^{9} b/s; 1990s: 10^{10} b/s. transmission bandwidth may increase even more in the future, particularly considering the potential of fiber optic technology. If the above empirical rule were to hold the data rate of the next few decades will be: year 2010: 10^{11} b/s and year 2020: 10^{12} b/s.

There is a significant trend toward outsourcing in the telecommunications industry. Suppliers in North America currently deal with numerous contractors and are finding that they are losing control of the costs and the quality of work. More and more, carriers are realizing the benefits of outsourcing to third parties so as to offer services such as customer-care and billing systems, network planning, and construction and operations support systems (OSSs).

Currently, 60 percent of service providers are outsourcing to third parties, but that number is projected to increase to 74 percent within two years. At present, 28 percent of service providers report that they outsource network planning and construction (i.e., deployment of the network) to third parties, and this number is projected to increase to 38 percent in the near future.

Data networks are altering the makeup of today's networks; as a result, suppliers in North America are expanding their networks to provide greater broadband to their customers. This is usually accomplished in one of two ways: through new construction or by retrofitting existing networks. Other trends include placing fiber and remotes closer to the home and upgrading switches. As a result, there are not enough installers for these areas, and existing installers cannot handle the peak load.

As the workload increases, the demand for quality installation and construction services increases as well. Using a large company with a permanent/long-term employee base as a single source supplier will ensure that the skilled manpower will be available to complete the project to the highest quality standards while controlling costs and schedule.

1.4 Understanding the Process

To appreciate the value of using a single source provider with project-management capabilities, one must understand all of the components involved in deploying a telecommunications network.

A simple analogy to describe the process for deploying an integrated telecommunications network today is that of a new subdivision that has just been developed. Within this subdivision, 500 new homes must be connected to the existing telephone network in the community. This example assumes that a remote-access node will be built within the subdivision. The process of connecting this subdivision to the local-access network switch, which connects to the long-distance access switch, which, in turn, connects to the rest of the world, is a multiple-step process (see Figure 1.3).

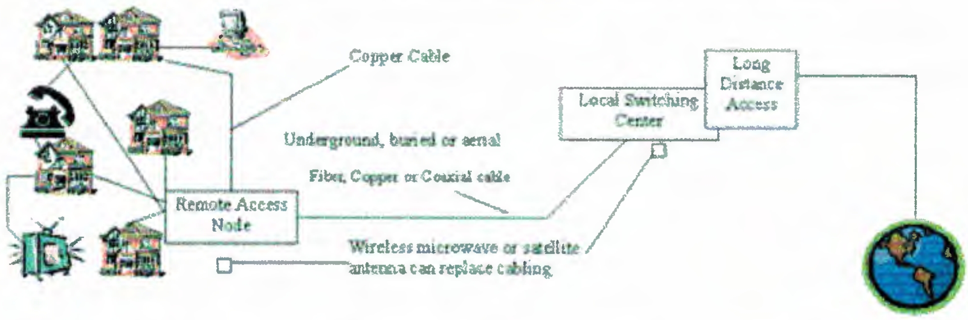


Figure 1.3 Example of Building an Integrated Telecommunications Network for a Subdivision.

Step 1: Outside-Plant Network Design (Engineering)

In keeping with the analogy mentioned earlier in the tutorial, the outside-plant designer for the project is responsible for route selection between the houses in the new subdivision and any reinforcement to the local switching center. In the design, the outside-plant designer can plan for the placement of underground, buried, aerial, submarine, or building cable installation or wireless installation, depending on a number of factors including the terrain, existing infrastructure, environment, etc. Specifically, the network designer is responsible for the following:

- Route planning
- Identifying right-of-way requirements and potential design conflicts
- Negotiating right-of-ways
- Determining specialized design, plan, and digital mapping requirements
- Preparing preliminary designs based on clients' specifications
- Developing firm price quotations based on preliminary design and estimating tables
- Providing as-built plans and specifications
- Completing final design and specifications for installation and ongoing design changes as required during installation

- Identifying material requirements and providing material ordering input to client or external suppliers as required

Step 2: CO Design (Engineering)

The design of the CO involves understanding what equipment must be installed to make the network work. Keeping with the subdivision analogy, we are assuming that there will be a remote-access node in the subdivision. In this particular example, the designer will be responsible for the following:

- Determining what equipment must be added to the existing switching center
- Determining the sizing of the access node
- Designing the transport system between the remote and the host
- Determining the digital equipment and transport system that will be used between the switch and the remote
- Reviewing the power system to see if it must be reinforced (power study)

Step 3: Outside-Plant Construction

Once the design and specifications have been determined, the installation of the cable (copper, fiber, or coaxial) must be installed to those specifications. This work is generally performed by highly skilled splicing and line technicians who are qualified to place and connect cable in a variety of outside-plant networks, including live circuits. Their responsibilities also include testing continuity and troubleshooting in existing networks. If it is determined that wireless technology is to be used, the infrastructure (i.e., towers) are built at this stage.

Step 4: CO Equipment Installation

The next step is to install and commission the specialized equipment to make the whole network work. Most CO and switching equipment is housed in a localized switching center or access nodes (remote switch) that is located within the subdivision that links back to the switching center. The CO equipment (COE) technicians are suppliers trained to install the specialized equipment that routes the calls to the appropriate switch. Some of the many types of equipment that must be installed and maintained by these technicians are as follows:

- Switching equipment including Nortel digital multiplex system (DMS) technology
- Transport equipment such as channel banks, fiber multiplex transport (FMT), digital access and cross-connect system (DACS), New bridge, and various miscellaneous peripherals (i.e., asynchronous transfer mode [ATM], frame relay, and network-management hardware)
- Access remote includes remote carrier urban (RCU) 600/900, DMS–1U, remote switch concentrator (RSC), and remote line concentrating module (RLCM), which are installed into various walk-in cabinets and environmentally controlled manhole enclosures
- FMS equipment patch panels, routers, bridges, and active hubs.
- Powers The technicians regularly install, replace, and upgrade rectifiers, inverters, batteries, and mechanized frame administration (MFA) power plants. They also install grounding into COs, access nodes, and customer-owned telephone rooms that are required to meet the grounding standards.
- Synchronous optical network (SONET) transport—The COE technicians are also experienced in building large Internet protocol (IP) networks and are supplier-trained to do system lineup and test (SLAT), including software upgrades on live equipment and optimization. They are also trained in optical carrier–3 (OC–3), OC–3E, OC–12, OC–48, OC–192, access nodes, and access node express.

Step 5: Commissioning

The commissioning of the newly installed network involves testing to make sure the network is up to specifications before it is turned on. Once it is determined that everything is operating according to specifications, it will be integrated into the live network.

Chapter two introduces the main components of the telecommunications network, which are the voice communication and data communication, and it studies the transfer of signals, switching and signaling.

CHAPTER 2

TELECOMMUNICATIONS NETWORK

2.1 An Overview

This chapter describes the basic operation of a telecommunication network with the help of a conventional telephone. The operation of a conventional telephone, which is easy to understand, is used to clarify how telephone connections are built up in the network. We look at the subscribers signaling over the subscriber loop of the telephone network. The same main signaling phases are needed in modern data and mobile networks. We start with this simple service to lay a foundation for understanding more complicated services. In this chapter we divide the network into layers and briefly describe different network technologies that are needed to provide various kinds of services. Some of these, such as mobile and data networks and their services. One of the topics which is introduced in this project is the theory of traffic engineering, that is how much capacity we should build into the network in order to provide a sufficient grade of service for the customers.

2.2 Basic Telecommunications Network

The basic purpose of telecommunications network is to transmit user information in any form to another user of the network. These users of public networks, such as a telephone network, are called subscribers. User information many have many forms, such as voice or data, and subscribers may use different access network technologies to access the network, such as fixed or cellular telephones. We shall see that telecommunications network consists of many different networks providing different services, for example, data, fixed, or cellular telephony service. These different services are discussed briefly due to the next three sections and approximately full explanation due to the chapter. In the following sections we introduce the basic functions that are needed in any networks regardless of what services they provide.

The three technologies needed for communication through a network are:

- Transmission;
- Switching;
- Signaling.

Each of these technologies requires specialists for engineering, operation, and maintenance.

2.2.1 Transmission

Transmission is the process of transporting information between end points of a system or a network. Transmission systems use three basic media for information transfer from one point to another:

- Copper cables, such as LANs and telephone subscriber lines;
- Optical fiber cables, such as high data-rate transmission in a telecommunications network;
- Radio wave, cellular phones and satellite transmission.

In telecommunications network the transmission systems interconnect exchanges, and these transmission systems altogether are called the Transmission or Transport Network. Note that the number of speech channels (which is one measure of transmission capacity) needed between exchanges is much smaller than the number of subscribers since only a small fraction of them has a call connected at the same time.

2.2.2 Switching

In principle all telephones could still be connected by cables as they were in the very beginning of the history of telephony. However, when the number of telephones grew, it was soon noticed that it was necessary to switch signals from one wire to another. Then only a few cable connections were needed between exchanges because the number of ongoing calls was much smaller than the number of telephones; see Figure 2.1. The first switches were not automatic and switching was done manually on switchboards.

Automatic switches, known as exchanges, were developed in 1887 by Strowger. Then the witching had to be controlled by the telephone user with the help of pulses generated by a dial. For many decades exchanges were a complex series of electromechanical selectors, but during the last twenty years thy have developed into a software-controlled digital exchanges that can provide additional services. Modern exchanges usually have quite a large capacity, tens of thousands subscribers, and thousands of them may have an ongoing call at the same time.

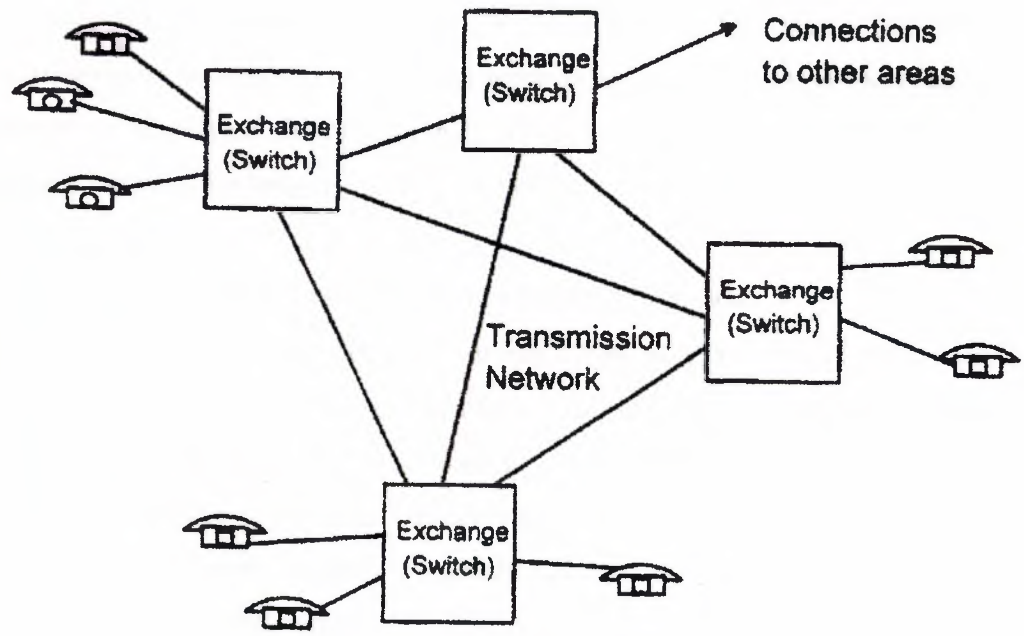


Figure 2.1 A Basic Telecommunications Network.

2.2.3 Signaling

Signaling is the mechanism that allows network entities (customer premises or network switches) to establish, maintain, and terminate sessions in a network. Signaling is carried out with the help of specific signals or messages that indicate to the other end what is requested of it by this connection. Some examples about signaling examples on subscriber lines are:

- Off-hook condition: the exchange notices that the subscriber has raised the telephone hook (DC-loop is broken) and gives a dial tone to the subscribers.

- Dial: the subscriber dials digits and that are received by the exchanges.
- On-hook condition: the exchange notices that the subscriber has finished the call (subscriber loop is connected), clears the connection, and stops billing.

Signaling is naturally between exchanges as well because most calls have to be connected via more than one exchange. Many different signaling systems are in use for the interconnection of different exchanges. Signaling is an extremely complex matter in a telecommunications network. Imagine, for example, a foreign GSM subscriber switching his telephone on in Hong Kong. In a few seconds he is able to receive calls directed to him. Information transferred for this function is carried in hundreds of signals of signaling messages between exchanges in international and national networks.

2.3 Components of a Telecommunications Network

In this section, we examine various types of networks. A telecommunication network can be viewed as an ensemble of a number of links, such as those shown in Figure 2.2. Telecommunication networks consist of three general categories of equipment: termination equipment, transmission equipment and switching equipment. Each of these three categories, in turn, comprises a number of subcategories or technologies. Figure 2.3 depicts a typical telecommunication network of the 1990s. As we can see, these networks can be very complex and many employ a variety of technologies.

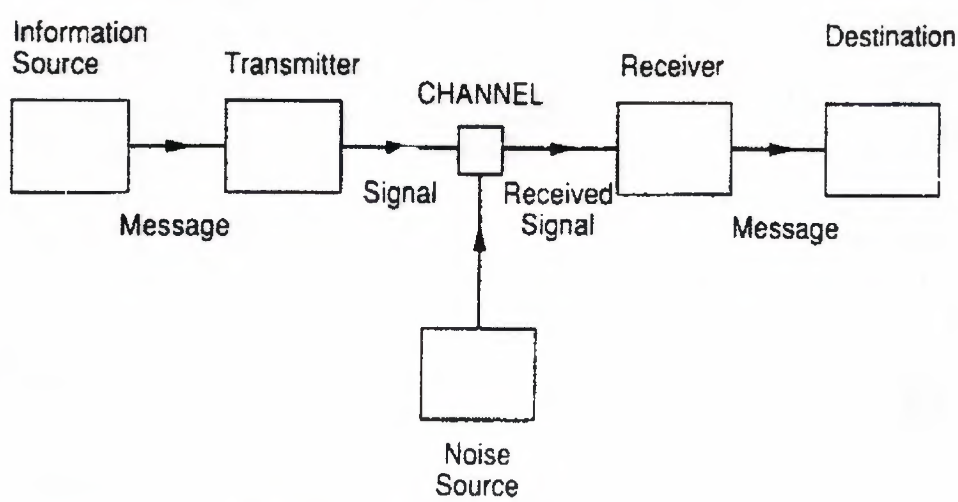


Figure 2.2 Communication Model.

Network can be public or private. Public networks can be used by everyone, similar to public transportation. Private networks are typically restricted to one organization (or a group of organizations); the general public is not allowed access to a private network, just as one can not employ (without permission or by arrangement) somebody else's private automobile. Both types of networks basically have the same functionality and capabilities, although different capabilities may be accentuated in one or the other. Public networks are provided by common carriers.

Public networks usually utilize carrier-provided switches, also known as exchange. Prior to the bell system divestiture, the distinction between exchange access and interexchange communication was not a matter of major regulatory importance, except possibly in terms of the rates and tariffs. After divestiture, terms such as "local exchange" and "interexchange" acquired an additional legal distinction; local exchange access service and interexchange service (more specifically, LATA-local access and transport area) must be provided by separate entities, and the old Bell companies are currently precluded from offering domestic interexchange services (with minor exceptions in some major corridors, for example, Northern New Jersey and New York City).

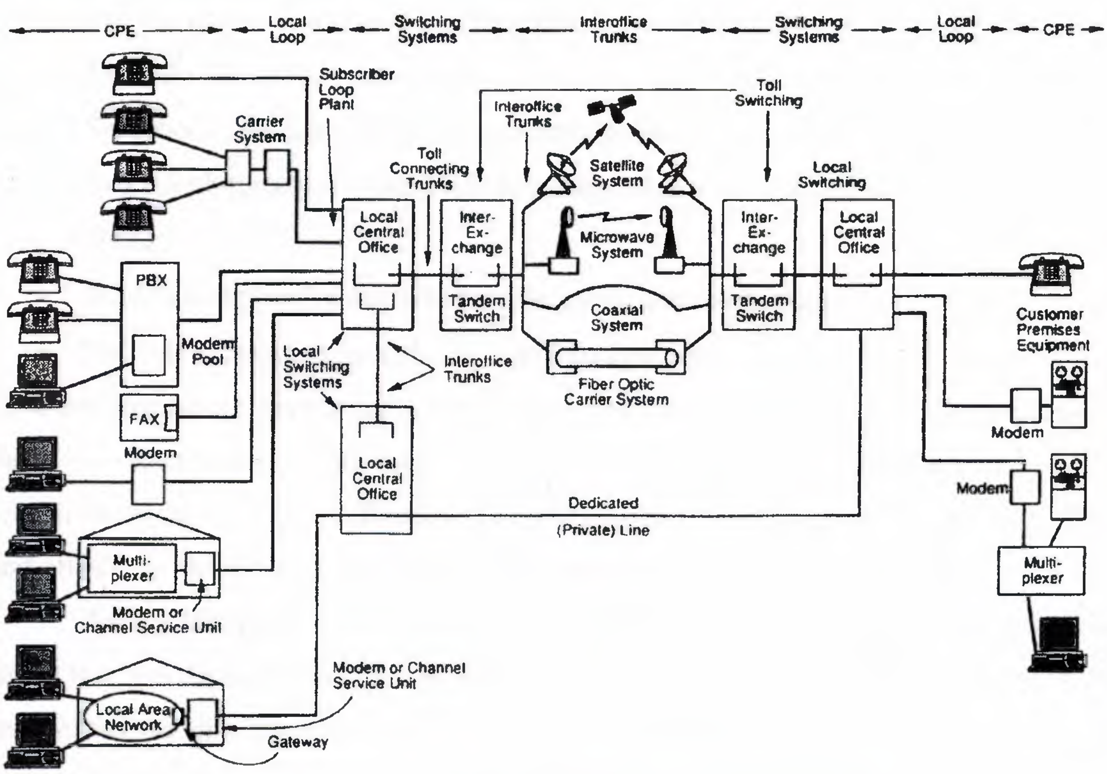


Figure 2.3 A modern telecommunication network.

2.3.1 Voice Networks

The traditional public switched telephone network was originally developed to service voice traffic. Data can also be carried by the same network when a modem (modulator-demodulator) is employed by users at each end of the link. In effect, the modem transforms the data into an acoustical signal that fits into the nominal 4-kHz bandwidth of a standard telephone channel. This method of carrying data is called voice band or circuit-mode data. Improved network facilities more suited to carrying data in their native digital mode are now beginning to emerge.

2.3.1.1 Customer-Premises Equipment

Customer-Premises Equipment (CPE) is equipment that is owned and maintained by the user. It includes the signal-entry termination equipment (called station in the voice environment), concentration equipment, in-building or in-campus wiring, or even an entire sub network. In the latter case, the user may own the communication network up to demarcation point (the point between the public network and the user's network), or in some cases the user can also own the long-haul transmission and switching equipment.

Typical CPE in the voice area includes:

- Telephone sets (particularly after computer Inquiry II, 1980, and divestiture, 1984);
- Key telephone equipment, including telephone sets, wiring, and other components;
- PBXs;
- Inside wiring (including wire closets, jacks, and connectors);
- Recording, answering, and voice mail equipment.

CPE termination equipment (as compared to other types of CPE) accepts the user's voice, data, or video signals and encodes them so that they can be transmitted over a telecommunication network. Examples of termination CPE include voice station set, CRTs and other data terminal, facsimile machines, and video conferencing cameras.

Analog termination devices encode the user signal (i.e., speech) in to an electrical signal that replicates the energy content of the original signal. This electrical signal is then transmitted to the desired remote location. Intuitively, digital termination devices "measure" the height of the signal at frequent intervals, and then represent that height with a binary coded number (more sophisticated methods perform signal analysis rather than simple energy measurement). For a computer terminal, the signal is already in digital form. The digital representation of the signal is then transmitted to the desired remote location, where it is utilized by the receiving CPE in digital form, or is converted into analog form. As an alternative, the user's CPE delivers an analog signal to the network; the network transforms this signal into a digital stream for more reliable transmission, and then reconverts it into an analog signal for delivery to the intended destination.

Binary numbers, employed in digital transmission, are composed of combinations of 0s and 1. The individual 0s and 1s are called bits. A collection eight bits is called octet (less precise term is byte). Sometimes octets and bytes are also called words, although the term is more appropriate to describe a collection of octet. For example a 32-nit computer is said to have 32-bit words (four octets). Digital communication channels are measured in terms of their information carrying capacity in b/s.

Initially, the public telephone plant was an analog network, optimized for analog voice transmission. This included analog transmission and analog switching facilities. Beginning in the early 1960s many interoffice trunks (links between switches) began to be replaced with digital links. In the mid-1970s, switches also began to handle digitized voice directly (i.e., without multiple conversions between analog and digital). Today digital links are common and prevalent. New network architecture, ISDN, aims at providing end –to-end digital circuits to the customer. All major telecommunication carriers in the United States and abroad have stated that ISDN is the strategic direction of their networks. Digital circuits are more suited to data transmission applications than are analog circuits.

(a) Voice Digitization Schemes

The telephone instrument performs the function of coding the user's voice into a signal suitable for subsequent transmission over the network. For public switched networks, an evolution is taking place in this area, tracking (sometimes leading) a similar evaluation in private networks. The evaluation is as follows. Until 1960s, the telephone set generated an analog signal, which was transmitted through the network in an analog fashion, end-to-end. Beginning in the 1960s and continuing through the 1980s, while the set still generated analog signal, the voice could be digitized in the transmission or in the switching components of the network {first in time in trunks between central offices (COs), 1962; then in the loops between the user and the CO, 1973; then in the proximity of the switch itself, so that a digital switch could terminate under multiplexed loop or trunk carrier systems, 1976}. With ISDN in the 1990s, the telephone set will be allowed to generate a digital signal representing the user's voice, for end-to-end transmission in digital form. (In a number of PBXs, this digitization at the telephone set is already taking place).

Historically, digitization techniques have been identified with activities performed in the network, particularly in reference to trunk carrier systems, digital loop carriers, and digital switches, as described in the previous paragraph. Because the future belongs to ISDN (or, at least, to digital communication), voice digitization techniques are here properly seen from a CPE perspective (the continued introduction of CPE high speed multiplexers also support this perspective). The digitization techniques do not change this change in perspective; the perspective only determines in what context the subject is treated. Remember, however, in spite of the present perspective, that as of 1990, a very large percentage of the voice digitization still occurs within the network proper.

To digitize the voice means to represent it with a stream of numbers coded in binary representation. Two classes of methods are used to digitize voice: waveform coding and vocoding. In waveform coding, we attempt to code and then reproduce the analog voice curve by modeling its physical shape. The number of b/s to represent the voice with this method is high: 64, 32, 16, or at least 9.6 kb/s, depending on the technology. Vocoding attempts to reproduce the analog voice curve by performing a mathematical analysis (fast Fourier transform) that "identifies" abstractly the type of curve; what is transmitted is a small set of parameters describing the nature of the curve. The number of kb/s to represent the voice with this method is low: 9.6, 4.8, 2.4, and even 1200 b/s, depending on the technology. Voice quality is increasingly degraded as the digitization rate becomes smaller. An extensive body of research on vocoding methods has evolved in the past 15 years (at least count more than 700 technical papers have been written).

Digital speech quality through a network is also degraded by the accumulation of quantization noise introduced at signal conversion points; conversion can occur several times in a network, in partially digital environment. In a nearly totally digital environment, only one analog-to-digital conversion close to the source and one digital-to-analog conversion close to the destination are required. In a totally digital environment, the conversion takes place right at the source, and not in the network. Voice quality will improve substantially in these environments, particularly with the high-quality coding.

(b) Pulse Code Modulation

The simplest waveform method to convert analog speech to a digital stream is process called pulse code modulation (PCM). PCM was invented in the 1930s, but only become prevalent in the 1960s when transistors and integrated circuits become available.

Nyquist theory specifies that to code properly an analog signal of bandwidth W with basic PCM techniques, we need 2W samples per second. For voice, band limited to a nominal 4000-Hz bandwidth, we need 8000 samples per second (the actual telephony frequency range used in 300 to 3400 Hz). The dynamic range of the signal {and ultimately the signal to noise ratio (S/N)} dictates the number of quantizing levels required. For telephonic voice, 256 levels suffice, based on psychoacoustic studies conducted in the 1950s and early 1960s, it follows that 8 bits are needed to represent many levels uniquely. This, in tern, implies that we need 64,000 b/s to encode telephonic human speech in digital form. PCM does not require sophisticated signal processing techniques and related circuitry; hence, it was the first method to be employed, and is the prevalent method used to day in telephone planet. PCM provides excellent quality. This is the method used in modern compact disc (CD) music recording technology (although the sampling rate is higher and the coding words are longer, to guarantee a frequency response to 22kHz). The problem with PCM is that it requires a fairly high bandwidth (64 kb/s0 to represent a voice signal. PCM is specified by the CCITT's (Consultative Committee on International Telephone and Telegraph) Recommendation G.711. The CCITT is a standards making body. Two "laws" (recommended standards) describe voice compression in PCM: in the United States, the μ-law is used; in Europe, the A-law is employed. The reason to follow specific PCM standards is that we want to be able to install equipment from different manufacturers and still retain system integrity and compatibility. (Although PCM can be mathematically treated as a type of modulation-which we will discuss later-many people today view it as an example of signal processing; this is the perspective we will use herewith).

One key issues is the spacing in the signal amplitude postulated by the sampling codec (also known as quantizer) to establish the boundaries where the different levels are declared. If we divide the maximum amplitude in 256 equal intervals, voice, which normally has numerous low-level signal components, would not be coded adequately. Instead, the amplitude space is sub divided with logarithmic spacing with respect to the signal origin, this affords a stable S/N ratio over a wide range of voice levels. Note, that if the input signal amplitude exceeds the maximum quantizer level, the result is clipping distortion. A quantizer must be designed to avoid frequent clipping; hence. The quantizer's maximum level is determined by the power of the strongest signal that the quantizer must handle. A signal-to-distortion ratio (S/D) of around 35 decibels is desired for a wide range of input level.

(c) Newer Coding Schemes

PCM has been around for a quarter century, and new technologies are beginning to demand attention. Sophisticated voice coding methods have become available in the past decade due to the evaluation of VLSI technology; 64 kb/s PCM is no longer the only available technique. Coding rates of 32,000 b/s, 16,000 b/s, and even "vocoder" methods requiring 4800 b/s, 2400 b/s, and even less, have evolved (intelligibility, but not speaker recognition, can still be obtained at 800 b/s).

Some interest exists in pursuing these new coding schemes because the implication is that we cam double or quadruple the voice carrying capacity of the network in place without the introduction of new transmission equipment. Of all available schemes emerging from the laboratory the adaptive pulse code modulation (ADPCM) schemes is the most promising at this time. It effectively provides "toll quality" voice with minimal degradation at 32 kb/s. the CCITT studied this algorithm and recommendation (G.721) followed in 1988. A problem with this method has been that of "passing" data at various speed under these coding methods; a number of widely deployed U.S. modems (in particular, the Bell 202 type, at 1200 b/s in half-duplex mode) fail to transmit through a digital carrier system equipped with the 32 kb/s line cards. Algorithmic refinements to deal with the problem involve fine-tuning some of the parameters that characterize the coding scheme. Performance of the coding scheme revolves around the following parameters: frequency response and tracking, idle circuit noise, transient response and warmup period, single frequency distortion, intermodulation distortion, and S/D. A standard for a 16 kb/s coding scheme and a proposal for an 8 kb/s scheme has been studied by CCITT study group XV. A brief description of the ADPCM follows.

(d) Differential PCM

If a signal has a high correlation (exceeding 0.5) between adjacent samples, as is the case for speech sampled at the Nyquist rate, the variance of the difference between adjacent samples is smaller than the variance of the original signal. If this difference is coded, rather than the original signal, significant gains in S/D performance can be achieved (conversely, a given S/D can be achieved with fewer quantizer bits). This implies that, for the same desired accuracy, fewer bits are needed to describe the change value from one sample to the next than would be needed to describe the absolute value of both samples. This is the idea behind differential PCM (DPCM). DPCM systems are based primary on a 1952 patent by Cutler.

In a typical DPCM system, the input signal is band-limited, and an estimate of the previous sample (or a prediction of the current signal value) is subtracted from the input. The difference is then sampled and coded. In the simplest case, the estimate of the previous sample is formed by taking the sum of the decoded values of all the past differences (which ideally differ from the previous sample only by quantizing error). DPCM exhibits the greatest improvement over PCM when the signal spectrum is peaked at the lower frequencies and rolls off toward the higher frequencies.

The problem with this voice coding method is that if the input analog signal varies rapidly between samples, the DPCM technique is not able to represent with sufficient accuracy the incoming signal. Just as in the PCM technique, clipping can occur when the input to the quantizer is too large; in this case, the input signal is the change in signal from the previous sample. The resulting distortion is known as slope-overload distortion.

(e) Adaptive DPCM

In adaptive DPCM, the coder can be made to adapt to slope overload by increasing the range represented by the encoded bits, which here number 4. In principle, the range implicit in the 4 bits can be increased or decreased to match different situations. This will reduce the quantizing noise for large signals, but will increase noise for normal signals; so, when the volume drops, the range covered by the 4-bit signal drops accordingly. These adaptive aspects of the algorithm give rise to its name. ADPCM transmits 4 bits per sample for 8000 samples per second, for a bandwidth of 32,000 b/s.

In practice, ADPCM coding device accepts the PCM coded signal and then applies a special algorithm to reduce the 8-bit samples to 4-bit words using only 15 quantizing levels. These 4-bit words no longer represent sample amplitudes; instead, they contain only enough information to reconstruct the amplitude at the distance end. The adaptive predictor predicts the value of the next signal on the level of previously sampled signal. A feedback loop ensures that voice variations are followed with minimal deviation. The deviation of the predicted value measured against the actual signal tends to be small and can be encoded with 4 bits. In the event that successive samples vary widely, the algorithm adapts by increasing the range represented by the 4 bits through a slight increase in the noise level over normal signals.

(f) Lower Rate Voice Digitization

Some CPE equipment (for example, T1 multiplexers) now use continuously variable slope delta (CVSD) to achieve voice digitization rates below 32,000 b/s. to understand CVSD, we consider a form of DPCM where the length of the digital word per sample is a single bit. With such a small digital word, more samples compared to PCM-DPCM can be sent in the same bandwidth. Clearly, 1-bit words can not measure loudness; hence, rather than sending the change in height of the analog signal curve, the 1-bit CVSD data refer to a change in slope (steepness) of the analog signal curve. At the sending end, CVSD compares the input analog voltage with a reference voltage: if the signal is greater than the reference, a “1” is sent and at the same time the slope of the reference is increased; if the input is less than the reference, a “0” is sent and at the same time the slope of the reference is reduced. CVSD attempts to bring the reference signal in line with the incoming analog signal. The steeper the slope (positively or negatively), the larger the output changes between samples; the CVSD algorithm increases the size of the step taken between samples each time the slope change continues in the same direction. This is similar in concept to the adaptive nature of ADPCM: a series of 1s produce a progressively larger increase in the output.

The receiver will reconstruct the sender's reference voltage, which, after being filtered, should be a replica of the original input. While one normally employs the CVSD at 32 kb/s, the actual digitization rate can be selected by the user in some systems (with ensuing voice quality implications). CVSD can operate from 64 to 9.6 kb/s. the quality deteriorates as the bandwidth decreases; this is the method employed by systems that provide 16 kb/s voice rates (the speaker still recognizable at 16 kb/s, and speech is still intelligible at 9.6 kb/s). A typical equipment line card converting four analog voice channels into four 16 kb/s digitized speech streams and multiplexing them onto a 64 kb/s channel costs around $4000 in 1990.

(g) High-Quality Telephony

Recently, the CCITT recommended an international standard (G.722/G.725) for coding wideband speech and music (50 Hz to 7 kHz at 3-dB attenuation) at 64 kb/s. this frequency range, extended at both the high end and the low end, considerably improves telephonic voice quality over the existing norm, approaching the quality of a typical car's FM radio. Extending the cutoff frequency from 300 to 50 Hz improves the naturalness of the audio signal. Some applications for these coders are in high-grade telephones in ISDN and for teleconferencing applications. In audiovisual conferencing applications, we would like to approach the quality of face-to face communication.

The codec performance requirements for voice band data are substantially different from those of voice signals. If the codec is required to encode voice band data signals as well, its cost and complexity increase. If the codec is not required to carry data, it can be optimized for best performance on speech signals. Subband coding techniques separate the signal into components occupying contiguous frequency bands, and encode the components separately. With the audio signal subdivided into two 4-kHz bands, a high S/N in the lower band becomes perceptually more important than in the higher band. An advantage of a design that uses two equally wide subbands is that each component can be subsampled to 8-kHz and the total transmission rate may be reduced in 8 kb/s steps by reducing the number of bits assigned to samples in one or the other band. A typical wideband coder accepts a 16-

kHz sampled input signal and split it into two 4-kHz (80 kHz sampled) bands using a quadrature mirror filter. The two 8-kHz sample rate subband signals are then encodec using an ADPCM coder. The upper band is encoded using 2 bits, while the lower band is allocated various bits, depending on the desired overall rate: 6 bits are used for 64 kb/s, 5 bits for 56 kb/s, and 4 bits for 48 kb/s. the two lower rates allow simultaneous transmission of an 8 kb/s (6.4 kb/s for data and 1.6 kb/s service channel) and 16 kb/s (14.4 kb/s for data and 1.6 kb/s service channel) data stream, respectively, in addition to the voice, on a signal 64 kb/s (ISDN) channel.

(h) Voice Digitization Summary

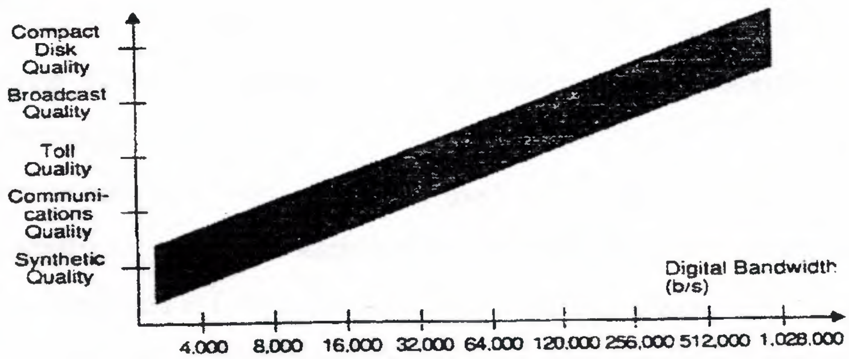


Figure 2.4 Digital voice qualities as a function of the digitization rate.

Figure 2.4 summarizes the quality versus digitization rate relationship associated with various voice coding schemes.

2.3.1.2 Transmission Equipment

The process of moving information from point to another is called transmission. The undertake transmission, one needs a variety of facilities. Transmission equipment typically includes terminal equipment (not to be confused with CPE termination equipment), which accepts the user's signal and changes it appropriately, and an interconnection medium, such as copper wire or coaxial cable among others. To superimpose the user signal onto the medium, the transmission equipment needs to modulate a carrier signal. Additionally, the equipment may multiplex a larger number of users over the same physical medium. (Multiplexers can also be CPE, if desired or appropriate).

(a) Multiplexing Schemes

A number of multiplexing schemes are available to place multiple calls in a standardized fashion on one medium. The basic multiplexing schemes are:

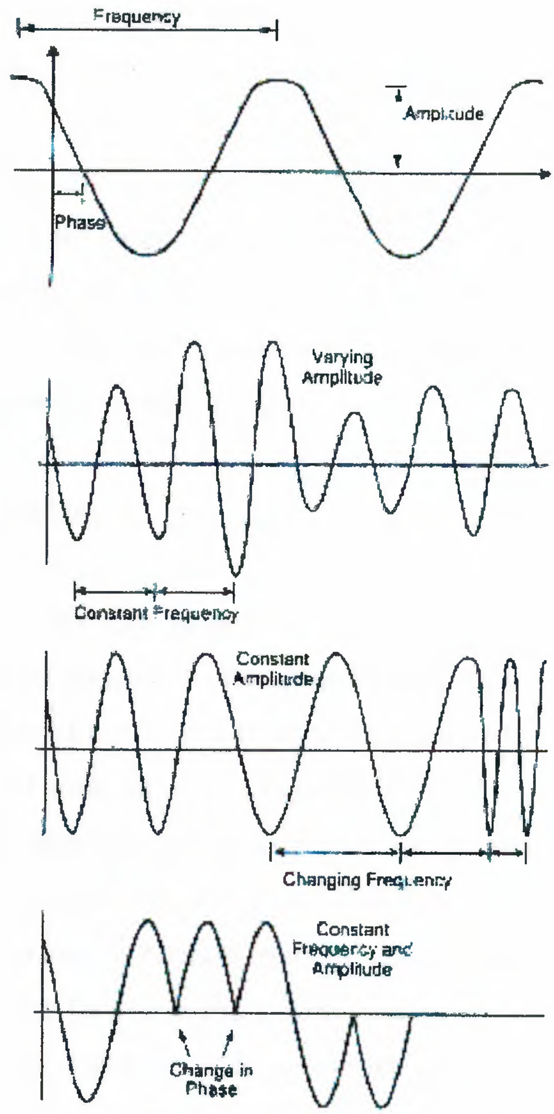
- Frequency division multiplexing (FDM). This is atypical of analog coaxial, microwave, and radio systems.
- Time division multiplexing (TDM). This is typical of digital transmission; it lends itself well to computer interfaces. In the traditional telephone network TDM has been used in conjunction with PCM coded signals.
- Space division multiplexing. An example is the frequency reuse in a cellular system or satellite.
- Code division multiplexing. Systems where the multiplexing is achieved by employing different data-stream coding methods. Used principally by military communication systems.
- Random access techniques. A method used in conjunction with TDM in which the multiplexing occurs via statistical bid and assignment of the channel. This is typically employed in LANs that do not use token disciplines.
- Demand assignment techniques. Bandwidth reservation-based systems; used in conjunction with random access. This approach is typical of satellite systems.

(b) Modulation

At the functional level, modulation is the process of imparting an intelligent signal onto an underlying carrier signal so that it can then be transmitted over a distance. The carrier signal depends on the media at hand (copper, microwave, fiber, etc.). Modulation is very common: radio and television, to mention only two obvious communication systems, employ modulation. Modulation functions come into play across the network for all types of transmission systems. The function of the modulator is to match the encoder output to the transmission channel. Figure 2.5 depicts the three characteristics of a electrical sinusoidal carrier, typical of media such as copper, coaxial, and radio: the amplitude, the frequency, and the phase. All three of these factors can be affected in order to achieve modulation. The three types of modulation are: amplitude modulation, frequency modulation, and phase modulation. The change to the carrier under each of the three methods is also shown in Figure 2.5. The modulating signal (the intelligence to carry) can be analog or digital.

Amplitude Modulation. A carrier's amplitude represents the instantaneous strength of the signal, as depicted in Figure 2.5. A carrier's amplitude of the modulated (AM), varies according to the amplitude of the modulating signal, while keeping the frequency constant. The modulation process produces a power spectrum that is symmetrical with respect to the frequency of the carrier: when viewed in the frequency domain, the modulated signal will have power spectral lines at sums and differences of the carrier frequency with the frequencies of the modulating signal. Although a major potion of the transmitter's power remains at the carrier frequency, amplitude modulation shifts energy into the sideband frequencies. This energy in the sidebands is what allows the remote end to demodulate the original intelligent signal.

Signal-sideband (SSB) modulation, an improvement of AM, concentrates most of the energy of the transmitter into the intelligence-bearing portion of the signal, enhancing the receiver signal. Because the upper and lower sidebands in AM modulation contain redundant information, one of the bands can be suppressed, after the modulation stage, giving rise to a "single sideband". This differs from AM, which uses transmitter energy to feed the carrier frequency and adds little to the intelligence received at the far end, as indicated above. SSB modulation results in a signal that requires reduced transmission bandwidth, in effect, allowing more intelligence to be transmitted over the same channel. The carrier signal can also be partially suppressed so as to use less power. This technique is often employed in microwave systems.



Top: Three characteristics of a carrier signal, Second from top: Amplitude Modulation, Third from top: Frequency Modulation, Bottom: Phase Modulation.

Figure 2.5 Modulation of the carrier

Frequency Modulation. Frequency modulation (FM) was developed in 1930s as an improvement over AM to provide high-quality music broadcasting. With FM, intelligence is added to the carrier wave by varying the frequency of the carrier in step with the frequency of the intelligence signal, while holding the output power of the carrier constant. FM is more immune to major sources of noise than AM because the most common type of noise tends to affect the amplitude. However, a frequency-modulated signal needs more bandwidth as compared to AM; even narrowband FM requires nearly twice the transmitter bandwidth of AM. FM is still extensively today for analog microwave transmission systems.

Phase Modulation. Phase modulation (PM) is similar in some respects to FM. The phase the transmitter is increased or decreased in accordance with the modulating intelligence signal. However, because small changes in phase are difficult to detect, PM is not generally used for analog applications. PM is used more commonly in digital modulation, for data transmission applications.

2.3.1.3 Subscriber Loop Plant

CPE equipment normally is connected to remote locations through a telephone company's CO. the subscriber loop is the physical link by which customers are connected with the telecommunication network. In the United States, local loops are usually provided by local exchange carriers (LECs). A star topology is employed, with loops emanating from the central point, the CO, to all local users of the network. This not only facilities management and trouble shooting of the loops but also allows central switching. Loops may be discrete two-wire copper facilities from the CO to the user, or may be partially multiplexed over a few miles of common medium. (Local loops for special services, such as data communication, may be four-wire). In this context, transmission equipment consists of the underlying physical channel, such as twisted pair, coaxial cable, optical fiber, microwave, and others, and multiplexing equipment such as digital loop carriers. To gain a perspective, we note that long-haul circuit mileage is only around 10% of the total U.S. telephone plant circuit mileage; 90% is the local loop, where the LECs install nearly 200 million miles of copper wire a year.

The physical infrastructure supporting local loops is known as outside plant. The outside plant includes conduit, poles, cable protection devices, terminals, aerial drop wire, feeder cable, and distribution cable. The cable may be strung from poles, buried, or placed in conduit. The cable is often buried underground without conduit because this approach is usually less expensive than housing the cable in conduit. The local loop is the part of the circuit that traditionally has been most susceptible transmission impairments; therefore, design and testing are both very important to maintaining quality. In designing a loop cable's characteristics, loop resistance and capacitance of the pair must all be carefully considered by the telephone company's engineers. Today's telephone networks are built around carrier serving areas (CSAs), and are served by a combination of digital or analog transmission and switching equipment. The radius of CSAs was originally defined by the bandwidth-distance performance characteristics of copper; typically, this radius is 12,000 feet. Fiber systems may eventually change the CSA concept.

The local loop infrastructure is made up of two components: the feeder plant and the distribution plant. The feeder plant provides (but not in every case) what are called "carrier facilities"-multiplexing-transmission facilities in which a number of customers may be multiplexed onto a single transmission medium. The number of multiplexed channel can vary from 24 to 96, or even to 672 or more. The LEC can combine many user channels with the multiplexing equipment, and deliver the signal to the CO where it is demultiplexed and fed to the switch for further handling. (In newer switched, it can be fed directly to the switch without demultiplexing equipment). The advantage of multiplexing is that construction of the physical plant is typically the most expensive component of the telecommunication networks (often more expensive than the electric equipment itself); thus, one of the objectives is to minimize the number of discrete physical channels needed, two methods of multiplexing in the feeder plant are frequency division multiplexing (analog) and time division multiplexing (digital). Time division techniques are becoming prevalent because of the increased deployment of digital networks and the lower cost to achieve multiplexing compared to analog techniques. The equipment at the remote end of the feeder plant also typically provides the voice digitization function for the group of stations being served.

The early examples of the feeder plant (through the 1960s) did not necessarily involve the use of multiplexing equipment. The only shared facilities in this case would be underground conduit or the poles. The distribution plant would be put in place once, to reflect anticipated growth in the sector served; the resulting junction between feeder and distribution plant occurred at a pedestal box or pole-mounted cross-connect box. In the late 1950s and 1960s, carrier systems grew in importance, due to the increased cost-effectiveness of remote electronics; an objective was set to achieve 30% of the feeder plant using carrier. At the interface, several lines can be integrated onto a single link (fiber or copper carrier) through multiplexing. The economics of scale established in the feeder plant reduce wire and cable costs that outweigh those of multiplexing. In the contemporary context, the feeder plant is largely implemented using carrier systems, and an increasing proportion of all loops are multiplexed.

The distribution plant is that portions of the network between the feeder termination (normally, some type of channel bank or remote switching module a satellite CO) and the customer. The distribution plant is generally copper-based in interoperate with the existing generation of telephone sets. Existing telephone sets require a current to operate the bell; the equipment at the feeder termination typically provides the appropriate signaling to the stations over the distribution plant. Two-wire twisted-pair cable (sometimes four-wire) is used in the distribution plant. Ideally the distance between the users and the feeder plant interface should be optimized to reduce the total length of wire required.

Fiber facilities are being increasingly introduced in the feeder plant. The only interfaces of the feeder plant are the (digital) switch at one end, and the carrier electronics at the other end. The feeder plant is thus amenable to fiber replacement without requiring additional changes to the network. Cable ducts supporting the feeder plant are overcrowded, and therefore offer ideal opportunities for fiber. End-to-end fiber-based local loops are a prospect for the future. On the other hand, feeder lines are becoming the prime areas for introduction of fiber at this time, particularly because this will facilitate the introduction of broadband ISDN, and also narrowband ISDN to an extent. About three-quarters of all new fibers installation projects are now in the feeder plant. Numerically, the feeder market is larger than the long-haul (about 45% of the total by circuit mile). Eventually, fiber will also be deployed in the distribution plant, for a general customer, providing end-to-end fiber connectivity. At this time, many commercial customers are already being equipped with fiber local loops-the large office buildings, computer centers, etc. The deployment of fiber in the local loop (distribution plant portion) for small business and residential customers will probably have to wait until the mid-1990s, though numerous experimental trials have already been undertaken. Residential fiber will open up opportunities for new services, including video-on-demand, high-definition television

(HDTV), and other graphics services (HDTV provides large-screen projection with 35-mm slide quality).

As for advanced local loops, some trends can perhaps be extrapolated from the French experiment at Biarritz in the mid-1980s. The Biarritz city test bed offered 5000 homes a host of "futuristic" services on the all-fiber network, in addition to cable TV. These services were videophone (two-way video), sensing of facilities for remote home management, video databanks (movie libraries), HDTV, and computer services (the last three were originally scheduled for a future time). Initially, fiber loops for 1.4 million homes in France were to be installed by the end of 1987, and 7 million homes were to equipped by 1992 (estimated project cost: $10 billion). The plant was to be used for multiservice broadband applications, notably cable TV (which at the beginning of the trial period in 1982 was nonexistent in France). Instead of the traditional branch-and-tree architecture, which is typical of one-way cable TV distribution applications, the plan called for a star configuration to allow full-function two-way interactive services. Because of political, technical, cost, and user-acceptance problems, that target has now been scaled down: only half million homes will be completed by the early phase. The fiber demultiplexing equipment turned out to be more expensive than anticipated. Only 30% of the houses passed subscribed to the service; videophone in particular has not been very successful. The government promoters of the system are reevaluating the cost-effectiveness of the entire concept, and of fiber in particular. Hybrid coaxial systems are now being considered in an effort to reduce cost.

2.3.1.4 Switching Systems

From a user's perspective, the primary function of a voice telephone switch is to connect, on demand, telephone instruments or other properly configured CPE. Because it would be impossible (and unrealistically expensive) to have a direct line between every possible user pairing, switching systems were developed to achieve the needed interconnectivity at an affordable price. This switched network allows a user to connect with any other subscriber by dialing the subscriber's address, and it eliminates the need for point-to-point wiring. An important aspect of any CO is the dc voltage supply that is applied to all loops. No current flows in any of the loops until the switch hook contacts of a telephone set are closed. The dc voltage permits the telephone set to signal the CO by merely closing or opening contacts with the switch hook or rotary dial.

Switches are connected by interoffice trunks and must communicate with one another to set up an interoffice call. Originally, this conversation was undertaken in-hand, namely, within the same channel as the user's conversation; however, this process was prone to fraud and was slow and inefficient. Now many switches can communicate with one another in an out-of-band fashion using separate supervisory network called common channel signaling. During the 1990s, the deployment of this signaling network will become more extensive.

Modern switches contain a common control section, which manages the call connection process, and a switching matrix, which actually makes the connection possible. Four generations of switching systems have occurred: (1) step-by-step technology, (2) common control switching systems, (3) analog electronic switching systems, and (4) digital switching systems. Modern switches are in reality computers: stored program control (SPC) implies the ability to program the switch using software instead of having to add discrete hardware modules (as was the case in older switches). Digital switches are becoming prevalent, although many analog switches are still embedded in the telephone network.

A network may contain a hierarchy of switches, beginning with local switching systems, which are closest to the user, and then going higher via tandem switches to a regional or national switch. A five-level hierarchical infrastructure was used in the United States prior to divestiture, but other topologies have now emerged.

(a) Common Control Offices

Early switches dedicated to the facilities required to control a call to each call for the duration of the call. This was inefficient and also uneconomical; if a call was blocked by equipment in the use, the system was incapable of rerouting the call. A control method that eliminated these problems was required. Equipment introduced after 1940 makes use of a limited number of specialized shared equipment units to control the process. Much of the control function to direct the path that the call takes through the system is concentrated in a small number of pieces of equipment, and these are used repeatedly. A unit of control equipment performs its function on a call and then becomes immediately available to perform the same function on another call. This mode of operation is known as common control.

(b) Computer-Controlled Switching Systems

The common control hardware can be electromechanical or hard-wired electronic (as was the case until the early 1960s) or can be computerized (which was introduced in 1965). Beginning in the late 1950s, designers realized that if the hard-wired common control was replaced with a programmable computer, options could be supplied to users more easily. Additionally, new services could be provided at the switch. As mentioned earlier, these systems are referred to as stored program control. With SPC, the control of switching functions is achieved by instructions stored in a memory and new service features may be added by changing the contents of the machine's memory. Examples of such services include speed calling, three-way calling, call waiting, call forwarding, and others.

The advantages of computer SPC include:

1. Labor saving as a result of simpler administration (for the changing subscribers' information. And reduced maintenance effort;
2. Higher traffic capacity;
3. Space saving: we can replace an existing switch with a smaller one having a larger capacity;
4. Power saving;
5. Cost reductions due to continued VLSI cost improvements;
6. Flexibility to changes over the life cycle of the switch (which could be as long as 40 years); and
7. Economical offering of new advanced services to the subscriber.

The Bell system's first SPC switch was the number 1 Electronic Switching System (ESS), which went into service in 1965. Number 1 ESS was designed to handle the heavy traffic loads and high density of telephone customers in metropolitan areas. The same basic principles were used in the No. 2 ESS, introduced in 1970. The No. 2 ESS was intended for communities with local switching offices serving around 10,000 telephone lines. Three basic elements of this type of switch in its early manifestation were: (1) a switching matrix using high-speed electromechanical ferrite switches; (2) a control unit, which directs the operations and maintenance in the system; and (3) two memories: a temporary memory (call store) for storing information such as the availability of circuits, called number, calling number, and type of call; and a semi permanent memory (program store) containing all the information that the control unit needs to process the call and make a connection. The program store is semi permanent because it does not have to be changed as calls are processed by the system. Two control units and maintenance frame make up the control complex; switches are designed with two control units with lock-step full duplex processing so that no service degradation is experienced for active calls (or calls being processed) should a single processor fail. The various subunits that form a control unit include the program control, input-output control (of peripheral units such as switching networks), the call store, and the program store.

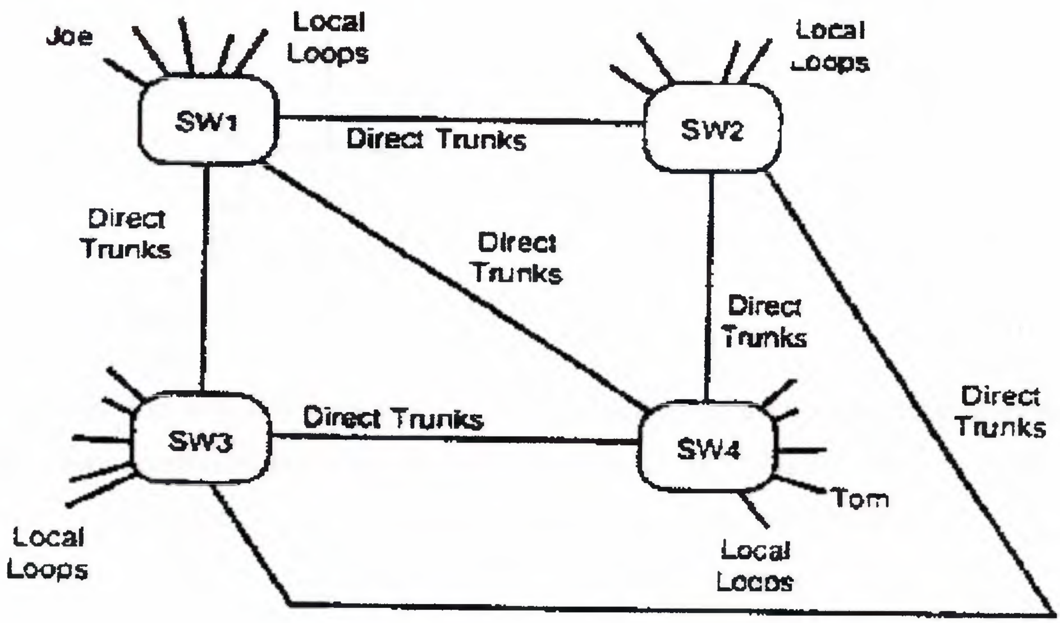
(c) Digital Switching Offices

Initially, switching was achieved by activating electromagnetic relays that would close to achieve a continuous metallic circuit end-to-end. This type of switching, also called analog, has a number of drawbacks, including unreliability due to mechanical components, noise added to signals due to the opening and closing of relays, large size inconsistency with digital transmission systems, and other drawbacks. Digital switching technology accepts digital signals and switches these to the desired destination by redistributing the signal electronically.

Voice digitization techniques have been widely applied, as indicated earlier, and their use is growing. Digitization is valuable in transmission because of its ability to protect the signal against the corruptive influences that degrade analog transmission. The use of corresponding digital carrier systems is increasing at such a rate (more recently because of the use of fiber) that to switch the digital signal directly is more advantageous than converting it to analog for space-division switching, then re-encoding it for transmission.

(d) Interoffice Trunks

Telephone switches are interconnected by a group of circuits called interoffice trunks. Typically these trunks are pooled, meaning that they can be seized by switches at either end and put on line as needed. The size of the pool is calculated in such a fashion that at the busy hour in the busy season no more than 1% (or some other small number) of the calls are unable to be routed to the next stage of the switching over the trunk pool system. Trunks are usually derived from carrier system of trunks, they are said to have direct trunks between them. (See Figure 2.6).



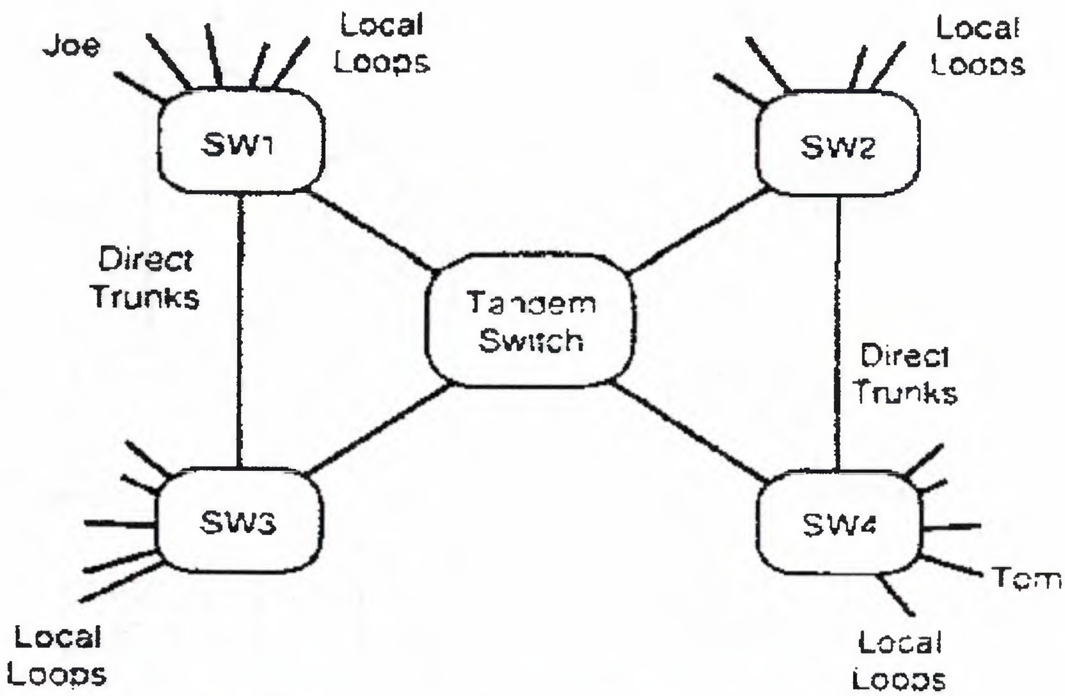
All witches are connected with trunks.

Joe talks to Tom over the trunks system between SW1 an SW4.

Figure 2.6 Direct Trunks.

(e) Tandem Switching Offices

In telephone networks with a large number of COs, having trunks between every pair of switches would not be practical. In these cases, the local telephone exchanges are connected to trunks that can provide access to an intermediate switching center, known as tandem switching offices, to which all switches are in turn connected. Connectivity between two local COs that are not directly connected by direct trunks must go through the tandem switch. Today, access to toll facilities is typically (but not always) via an interexchange tandem. (See Figure 2.7).



SW1 and SW3 as well as SW2 and SW4 are connected with direct trunks because of high traffic. Other connectors are made via the tandem switch. Joe talks to Tom over trunks to the tandem switch, the tandem itself and then over trunks to SW4.

Figure 2.7 Tandem arrangement.

(f) Interexchange Trunks

As a product of divestiture, long-haul communication is handled in almost all situations by a carrier other than one providing the local loops and the local switching system. A tandem arrangement may be employed to reach one or more interexchange carriers. (See Figure 2.8).

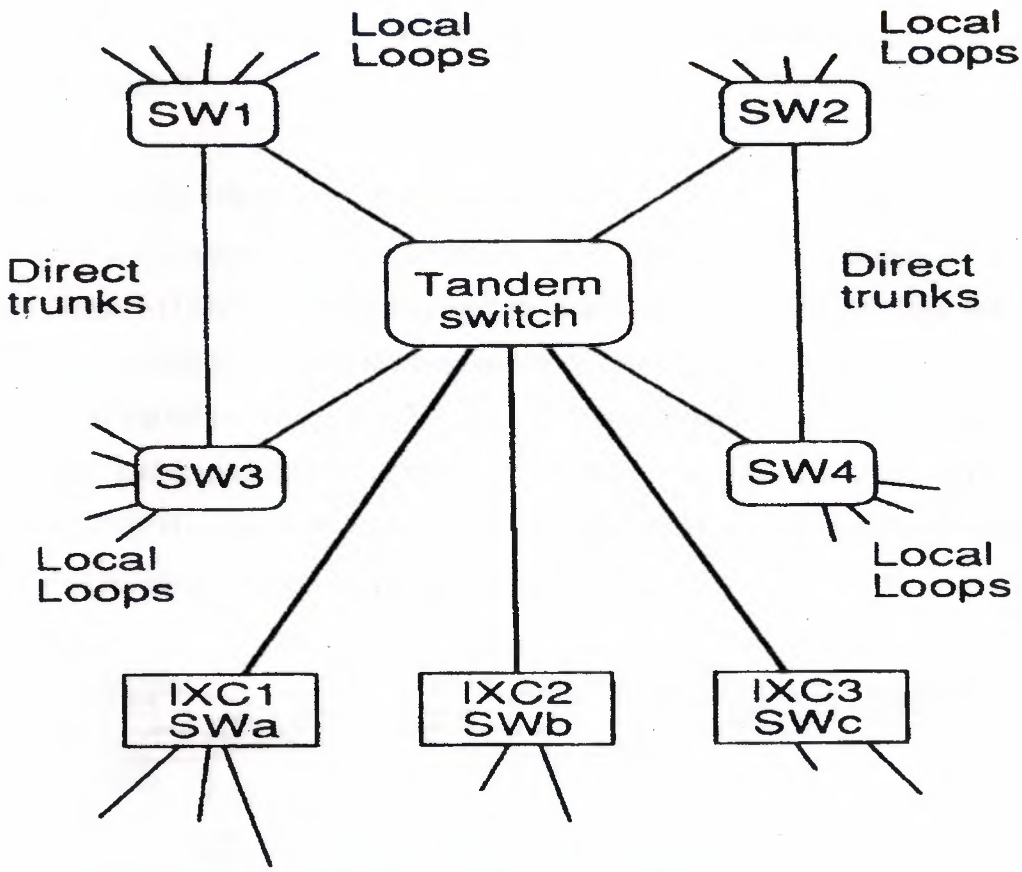


Figure 2.8 Typical access to long-haul carriers.

(g) Traditional Switching Hierarchy

A traditional tree-based hierarchy of switches ensures that there is a path from each switching office in the network to any other switching office in the network. It is an architecture, which had been employed by the Bell System for decades. A characteristic of the hierarchical structure of switching offices is that each office is connected to an office at a higher level except, of course, for those at the highest level; these top level offices are completely interconnected. The traditional network structure for the United States and Canada is divided into 12 regions; each region has one switching office at the highest level. The trunk group that connects a switching office to the next highest level switching office within a region is called a final group. Additional trunk groups supplementing the tree structure are permissible; in fact, they are desirable where sufficient traffic exists between switching offices not directly connected by the tree structure. These trunk groups, which are not part of the tree structure, are called high-usage groups. If one group is busy, a call will be rerouted to a different group until the final group is reached.

(h) Evolution of the Telephone Company Plant

Figure 2.9 depicts the five-stage evolution of the telephone company plant over the years. Stage 1 (1890s to 1950s) involved an all-analog plant. Stage 2 (1960s and early 1970s) saw the emergence of digital transmission. Stage 3 (mid-1970s and 1980s) saw the introduction of digital switching. Stage 4 (early 1990s) is experiencing the introduction of ISDN for true end-to-end digital connectivity. All these stages have involved voice band bandwidths (4000 Hz or 64,000 b/s); in stage 5 (late 1990s) we will see the introduction of end-to-end broadband digital communication.

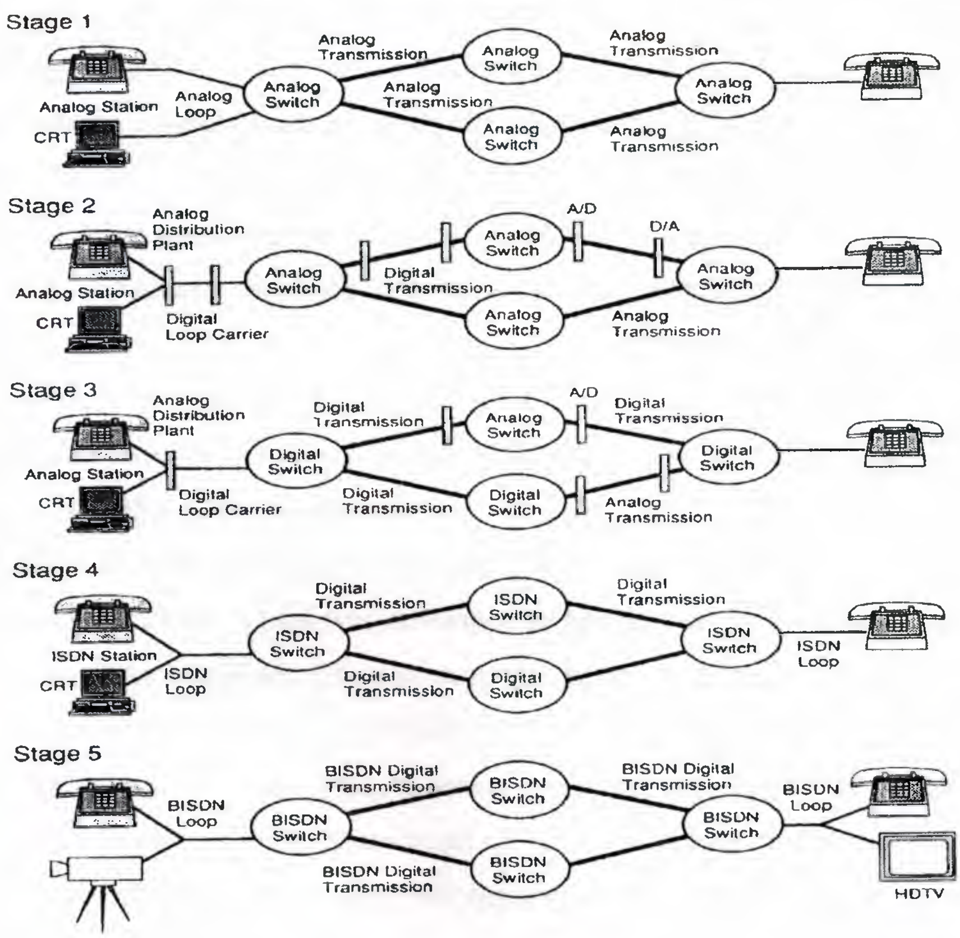


Figure 2.9 Evolution of Telco plant.

(i) North American Numbering Plan

Numbering schemes are mandatory for orderly identification of network subscribers. In traditional telecommunication networks, numbering plans are also used to accomplish routing.

A domestic toll voice call (with presubscription to an interexchange carrier) can be placed using an address such as 1-NXX-NXX-XXXX, where the first three digits after the 1 represent the area code, the middle three digits the exchange, and the last four digits the specific station connected to the switch specified by the NXX. A total of 160 area code combinations is possible under the current numbering plan, which requires the first digit to be a number between 2 and 9 (an "N"), the second digit to be a 0 or a 1, and the third digit to be between 0 and 9 (an "N"). code combinations 211,311,411,511,611,711,811,and 911, as well as 200,300,400,500,600,700,800, and 900 are reversed for special applications. When AT&T first assigned the initial area codes for the United States, Canada, and parts of the Caribbean in 1947, engineers projected that they would last 100 years. With a potential one billion phone numbers now assigned, the 100-year projection will fall short by more than 50 years. All currently available area codes will be assigned by 1995. The overall growth in telephone number usage is running at about 7 to 9% a year. The growth is fueled by growth in population, multiloop households, cellular telephony, paging, and facsimile machines.

Two recently introduction area codes are 708 in Illinois and 908 in New Jersey. Area codes that have been reserved include 903 (Dallas, late 1990); 510 (San Francisco, late 1991); and 310 (Los Angeles, a992). The remaining unassigned codes under the original numbering plans are: 210, 410, 706, 810, 905, 909, 910, and 917.

During the 1960s, a plan was developed to provide for the time when the existing area codes would be exhausted. The plan removes the requirement that the second digit of each area code be a 0 or a 1; eliminating this restriction makes 640 additional codes available. Implementation of the plan, however, is a major undertaking for all LECs. Each of the 792 COs in area code has a maximum of 10,000 telephone numbers associated with it, which means that each area code has approximately 7.9 million numbers than can potentially be assigned to customers.

(j) Traffic Engineering

The traffic offered to a switch is a function of two factors: the average rate of arrival of new call attempts and the average holding time of a call (assuming that the variance is small enough to be safely ignored). The averaging period for the origination rate is the busy hour, a one-hour period chosen to typify for a given CO the annually recurring hour during which the offered traffic load is a maximum. Peak busy hour calls is the unit used for expressing the processing capacity of a switching machine's control. The offered traffic load is expressed in hundred call seconds ("CCS" where the first C is from centum = hundred), and is the product of the number of calls and the average holding time, or the sum of the holding times of calls under consideration. By convention, the units of CCS are often used to mean CCS per hour. The average holding time multiplied by the number of calls placed per unit time is a measure of the traffic intensity and is often expressed in erlang, named after an early contributor to traffic theory, is the traffic intensity equivalent to one call held for an entire hour, and is therefore equal to 36 CCS per hour (it is also equivalent, for example, to 36 calls held for 100 seconds each in a one-hour period, or a circuit occupied 100% for a full hour by any number of calls). Put differently, the traffic intensity in erlangs is the number of channels that would be sufficient to serve a given offered load in a one-hour period, if the load timing could be rearranged so that all the channels would be continuously busy. It thus constitutes a lower bound on the number of channels to carry that traffic intensity.

2.3.2 Data Networks

While data networks differ in many ways from voice networks, the components identified above (CPE, switching, and transmission) are all still present. CPE may consist of CRTs (cathode ray tubes), computers, remote peripherals, *et cetera*. High-speed private networks, such as LANs, usually do not include an explicit tance or rejection based on the address posted on the message (the message is broadcast to all users of the network). Lower speed networks may involve circuit-switched or packet-switched facilities in the network, or switching in the host computer or associated front-end hardware.

2.3.2.1 CPE

CPE in the data environment includes:

- CRTs, printers, plotters, computers;
- Local area network (LANs);
- Private wide-area networks (WANs);
- Modems, multiplexers, channel service units; and
- Front-end processor.

(a) DTE and DCE

A data communication system consist of data terminal equipment (DTE), data circuit-terminating equipment (DCE)-colloquially known as data communication equipment-and the transmission circuit, also variously known as channel, line, link, or trunk. The DTE is a device, such as a terminal or a computer (mainframe, minicomputer, or microcomputer). The DTE supports end-user applications, for example, data entry, inquiry or response, and database management functions. The DCE provides the connection of the user DTE into the communication circuit. Notice that both DTE and DCE can be CPE (although some DCE may also be part of a public network). (See Figure 2.10).

A physical level specification (also know as a physical level interface), such as EIA 232-c, defines the following attributes of a data communication system:

1. The wiring connection between devices (when wires are used);
2. The electrical, electromagnetic, or optical characteristics of the signal between communicating devices;
3. The provision for mechanical connectors (dimension, number of pins, etc.);
4. The agreement on the type of clocking signals that will enable the devices to synchronize onto each other's signal;
5. The provision for electrical grounding (if needed).

For example, RS-232-c describes a standardized interface between DTE and DCE employing serial binary data interchange. RS-449 describes a general-purpose 37-position and 9-position interface for DTE and DCE employing serial binary data interchange.

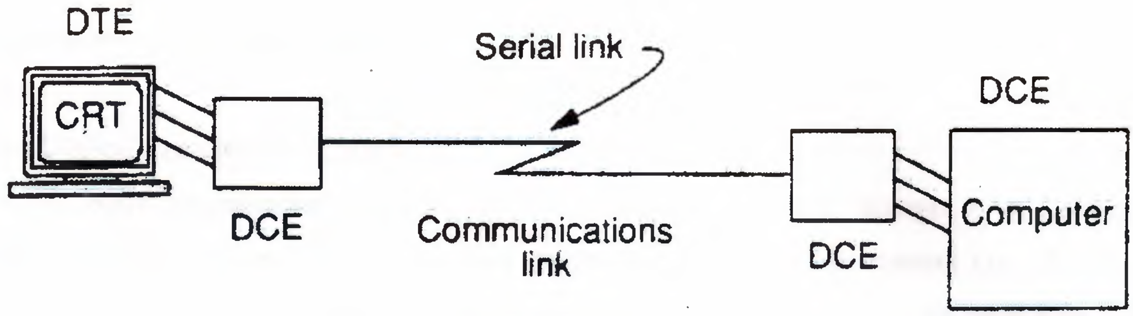


Figure 2.10 DTE and DCE.

(b) ASCII and EBCDIC Character Coding

One of the issues in data communication is the representation of the data, particularly when dealing with several computer systems. The applications in different systems may wish to represent data structures in different ways. Yet a common exchange structure is needed. Many computers use the 7-bit ASCII (American Standard Code for Information Interchange) code; other systems use the 8-bit EBCDIC (Extended Binary Coded Decimal Interchange code).

2.3.2.2 Transmission

The transmission link can be analog or digital. It can also be point-to-point or multipoint. Additionally, the link can also be characterized as being two-wire or four-wire, and half-duplex or full-duplex. These concepts are summarized below.

(a) Analog Transmission Methods

Analog signals vary in time in terms of amplitude and frequency. As indicated earlier, at the current time, analog network interfaces are still very common. To send data over an analog line we use a modem. The modem takes a digital signal and produces a signal suitable for transmission over the analog network.

The analog signal generated by a modem to transmit data consist of a carrier frequency, plus sidebands that change as the data bit pattern varies. These sidebands must fit within the attenuation limits of the voice-grade channel. If the timing will be distorted when reproduced by the distant modem. To minimize the effects of channel attenuation, the modems may include or require conditioning filters to produce an attenuation curve which complements that of the channel.

(b) Digital Transmission Methods

New transmission facilities, specially designed for data, accepts a digital signal directly, although a network termination device is still required to connect the CPE to the public network. The internal transmission method, however, is still based on analog media.

(c) Types Of Circuits

Point-to-point and Multipoint Data Circuits. A data circuit can also be classified as point-to-point or multipoint. A point-to-point circuit connects two devices only and a multipoint circuit connects more than two devices.

Two-wire versus Four-wire. The communication channel on the line side of the DCE is usually described as "two-wire" or "four-wire." These terms are derived from copper-based telephone nomenclature, in which two or four wires (or equivalent) are employed to transmit information. Four-wire affords more flexibility, but requires more transmission resources (i.e., cable). Typically, one pair of wires is used for transmission of information in each direction, and higher effective bandwidth is achievable. With two wires, more sophisticated electronics is required to achieve intelligible simultaneous two-way data transmission. The majority of traditional local loops for voice use employ two wires to minimize the investment in copper, as discussed earlier. Long-distance voice circuits employ four wires, so that a separated transmission path is used for each direction. The interface between the two configurations is provided by a device called a "hybrid."

Data circuits that need to carry 9.6 kb/s or higher on a sustained basis have traditionally consisted of dedicated transmission facilities (also known as private lines), which are four-wire equivalent end-to-end. Dedicated four-wire circuits are particularly common in multipoint applications. Two-wire, Full-duplex. Echo-canceling, error-correcting, high-throughput modems for use with switched lines are increasingly becoming available.

Half-duplex versus Full-duplex. These terms describe how data are transmitted across the channel, regardless of the physical configuration of the channel. Half-duplex defines a transmission in which data are transmitted in both directions but not at the same time. The DCE alternates between transmitting and receiving the data between the devices. Full-duplex describes the simultaneous transmission of data in both directions. The term “two-way alternate” is also used to describe a half-duplex data flow and “two-way simultaneous” to describe a full-duplex data flow.

The ability to operate in a full-duplex mode depends on: (1) the channel (i.e., if the echo suppressor found in a long-haul dialup circuit can be disengaged); (2) the modem; and (3) the protocol used by the communicating entities. If any one of these three components fails to operate in a full-duplex mode, then the overall transmission becomes (effectively) half-duplex. Today, the majority of the data communication systems are full-duplex.

(d) Line Turn-Around

When the DCEs use half-duplex schemes, a time interval is required for the devices to stabilize and adjust to the signal before transmission in the other direction. This stabilization period is called “training time,” and the process of reversing the signal is called “line turn-around”. Turn-around times of 100 to 200 ms are not uncommon. Because of this delay, most multipoint systems keep the master device’s carrier on constantly (constant carrier) with the slaves configured for switched carrier operation (switched carrier). This approach eliminates half the turn-around delay. Some DCEs have used split channel techniques to eliminate the switched carrier delay completely.

(e) Asynchronous and Synchronous Transmission

Two methods for formatting and transmitting user data through the channel exist: asynchronous and synchronous. The asynchronous approach is an older technique; however, due to its simplicity and relative low cost, it is still a common method and is found in many modern systems such as personal computers. Asynchronous transmission is characterized by the use of timing bits (start and stop bits) enveloping each transmitted character. The purpose of the start and stop bits is to provide for character synchronization and timing between the transmitting device and the receiving device. The stop bit notifies the receiving device that a character is being transmitted on the channel. The stop bit indicates that all the bits have arrived and provides for each other's timing functions. In asynchronous transmission, each character is framed by start and stop bits.

In the synchronous approach, all characters are directly blocked together and are transmitted without the intervening start and stop bits. Framing codes (called syncs or flags) are placed in front and behind the full data unit (usually called a frame) to indicate to the receiver where the user data begins and ends. Synchronous transmission can provide timing signals by one three techniques:

- A separate clocking line. A separate clocking line is a technique used for short-distance nontelecommunication connections: in addition to the data line, another line transmits an associated timing signal, which is used to clock the data into the receiver. For example, the EIA-232-D and CCITT V.24 specifications provide several options for synchronous transmission and clocking. Separate clocking channels are not practical for longer distances because the installation of a separate link or wire is expensive. Longer distances also increase the probability that the clocking line will lose the synchronization with the data line because each line has its own unique transmission characteristics. The telephone network does not provide clocking lines.
- Embedding the clock signal in the data stream with the data acting as a clock to a simple receiver circuit. To embed the clocking signal, the data bits are encoded at the transmitter to provide frequent transitions of the channel.
- Embedding the clocking signal in the data stream and using it to synchronize a receiver clock. The line transitions in the incoming bit stream keep the receiver clock aligned (synchronized) onto each bit in the data block.

(f) Digital Modulation

The principle of digital modulation are the same as those of analog modulation discussed earlier, except that the modulating signal is digital. In a digital environment, the modulator accepts binary encoded symbols and produces waveforms appropriate to the physical transmission medium at hand, which is always analog with today's technology. Because these signals involve discrete jumps from one state to another, the modulating action is generally described as keying.

The simplest technique, amplitude-shift keying (ASK), modulates the carrier with the binary signal to produce an AM signal. Binary 0 is represented by no amplitude of the carrier and binary 1 with full amplitude of the carrier. The off-on keying is simple, but ASK makes inefficient use of transmission power. Amplitude modulation is not often used by itself because of transmission line power problems and sensitivity to line errors. However, it is commonly used with phase modulation to yield a method superior to either FM or AM.

Frequency-shift keying (FSK) is also common. This simple FM technique uses the binary signal to switched between two frequencies. In its simplest form, it is characterized by abrupt phase changes at the transition between one state and the other, which distorts the spectral energy and reduces spectral efficiency.

Phase-shift keying (PSK) is also being increasingly used because of higher efficiencies achievable with this technique (packing more data onto the carrier, or "more bits per baud," in engineering jargon). Simple PSK uses the binary signal to alternate the phase of carrier between 0 degrees and 180 degrees. A variation, known as differential PSK, compares the phase of the previous bit with the present bit to determine if a transition has occurred.

Efficiencies of around 1 b/Hz of bandwidth are achieved with basic PSK, FSK, and ASK. Two bits per hertz can be obtained through the use of 4-phase PSK and quadrature amplitude modulation (QAM) (QAM is discussed below). More complex schemes, such as 16-phase PSK and 16-state QAM, are needed for 3 b/s per hertz. Four bits per hertz is achieved with 64 QAM. However, as the complexity of the modulation techniques increases, a better S/N is required to achieve the same bit error rate (BER) grade of service. (To be more precise, we mean bits per baud, which are changes in Signal State; for example, the V.32 modem operating at 9.6 kb/s has a modulation rate of 2400 baud and encodes 4 bits/baud).

(g) The Modem

As indicated earlier, the modem is responsible for providing the required translation and interface between the digital environment and an analog communication link. Modems are DCEs, and have become CPE in the United States. The modem provides a digital-domain to voice-domain conversion; modems should not be confused with the analog-to-digital process discussed earlier under the PCM technique. Modems are designed around the use of an analog carrier frequency in the voice band domain. The carrier is modulated with the DTE's data stream. The carrier signal is changed back to digital at the receiver DCE modem by the process of demodulation. Modems can use amplitude, frequency, and phase modulation, or a combination thereof to impart the data signal over the analog carrier. Each method impresses the data on a carrier signal, which is altered to carry the properties of the digital data stream.

Amplitude modulation modems alter the carrier signal in accordance with the modulating digital bit stream, while the frequency and phase of the carrier are held constant (the carrier is turned on or off, or at least changed in magnitude).

Frequency modulation modems alter the frequency of the carrier in accordance with the digital bit stream. The amplitude is held constant. In its simplest form, a binary 1 is represented by a specified frequencies and a binary 0 by another. The most common FSK modems use four frequencies within the useable bandwidth of a standard telephone circuit (nominally, the bandwidth is 4000 Hz). A typical full-duplex two-wire FSK modem transmits 1070-and 1270-Hz signals to represent a binary 0 (space) and binary 1 (mark), respectively. It receives 2025- and 2225- Hz signals as a binary 0 and binary 1. FSK has been a widely used technique for low-speed modem (up to 1200 b/s). It is relatively inexpensive and simple; many PCs use FSK modems. The most sophisticated modems employ multiple carriers, instead of two carriers.

Phase modulation modems alter the phase of the signal to represent a 1 or 0. A common approach to implement PSK is to compare the phase of the current signal state to the previous signal state. PSK techniques use bandwidth more efficiently than FSK, but they require more elaborate equipment for signal generation and data representation. Phase modulation techniques are almost exclusively used today on high-speed modems. Phase shifting can be used to provide multilevel modulation. Table 2.1 depicts a 4-phase PSK and an 8-phase PSK modulation scheme. In the former we can map 2 pits per baud (one 360-degree carrier signal rotation); in the latter, 3 bits per baud.

An extension of multiphase PSK modulation is quadrature amplitude modulation (QAM), in which the sine and cosine of the modem carrier frequency are amplitude modulated with two or more amplitudes. QAM techniques are widely used in high-speed modems. QAM is a combination of PSK with an amplitude for a set signal points.

Most modems use scrambling techniques to ensure a proper number of state transitions for accurate timing recovery at the receiver modem. The scrambling is usually done by the DCEs. Scrambling modem (DCE-to-DCE) synchronization. However, synchronization of synchronous transmission between user devices and modems (DTE-to-DCE) must be performed with a separate timing (or clocking) circuit.

Most modems that operate with speeds up to 4.8 kb/s employ fixed equalizers; these circuits are designed to compensate for the average conditions on a circuit. However, the fixed equalizers are being replaced with dynamic (or automatic) equalization: the modem analyzes the line conditions and adjusts its equalization accordingly. The adjustments occur very rapidly, on the order of 2400 times a second for a 9.6 kb/s modem. More information on equalization follows.

Table 2.1 Quadrature Modulations

4-PSK	8-PSK		
Bits to Code	Phase Change in Signal (deg)	Bits to Code	Phase Change in Signal (deg)
11	45°	111	22.5°
10	135°	110	67.5°
01	225°	101	112.5°
00	315°	100	157.5°
		011	202.5°
		010	247.5°
		001	292.5°
		000	337.5°

(h) History of Modems

The 70-year history of voice band modems falls four into phases:

1. From 1919 to the mid-1950s, work arose out of the need to transmit telegraphic information over the voice network. Research focused primarily on the basic properties of copper lines and on basic theories of data communication. The maximum data rate was around 100 b/s.
2. Starting in the mid-1950s, growing military requirements, and nascent commercial interest in transmitting large amounts of data, led to efforts to achieve greater transmission speeds. The technical investigation concentrated on modulation techniques, telephone line characteristics, and receiver design. Methods to deal with marginal phase distortion in additive noise channels, equalization, intersymbol interference, channel amplitude distortion, and delay distortion were developed. This resulted in an increase in the speed from 100 b/s at beginning of this period to 9600 b/s in the late 1960s. By the late 1950s, AT&T introduced the bell 103 (300 b/s) and bell 202 (1200 b/s) modems; these employed FSK principles. In the early 1960s, the application of 4-phase PSK modulation, resulted in the bell 201, which provided 2400 b/s over conditioned lines. Commercial products in the late 1960s provided reliable higher speed bandwidths; notable in retrospect were Milgo, which achieved 4800 b/s with 8-phase PSK, and Codex, which achieved 9600 b/s with QAM techniques with a 16-point signal constellation (i.e., 4 b/Hz).
3. During the 1970s, the speed on commercially available modems remained around 9600 b/s, but major design improvements led to significant reductions in size and power. Techniques implemented in this period, including LSI and VLSI, timing recovery, adaptive filtering, and digital signal processing, would establish the basis for the advancements of the next phase. Some improvements in speed (to 14,400 b/s) were obtained by using more advanced equalization techniques.

4. During the 1980s, error-correcting modems, advanced signal processing, and higher speeds were introduced-19,200 b/s is now routinely possible on dedicated lines. Speeds as high as 38.4 kb/s are now achieved using data compression. Because telephone lines use fiber for a larger for a large portion of their total span, they are less prone to some of the traditional problems, including phase jitter; S/N of better than 28 dB are achievable. QAM signal constellations with 64 points (6 b/Hz) become practical. The 19,200 b/s modems use orthogonal multiplexing (transmission of several noninterfering subsignals over the common channel), or multidimensional trellis-coded modulation, which we will discuss latter. In band-limited channels an increase in transmission rate requires an increase in the number of coding points in the constellation; this, however, can run into marginal-performance areas (in terms of signal quality) of the channel. An approach to dealing with possible mutilation of some constellation points is to use error-correction bits. Until the early 1980s, however, it was thought that the increased speed would wash out compared to the overhead needed to provide the needed error correction. This turns out not to be true. Trellis-coded modulation can improve the performance of a modem by 3 to 6 dB.

In spite of repeated predictions over the past decade that modems would soon be eliminated by end-to-end digital networks, the modem industry continues to prosper. While digital backbones are becoming popular, a large portion of data communication in 1990 is still carried by voice band modems over the analog telephone network. Modems are now available on one or two chips. Effective bandwidth has increased approximately 30-fold during the past decade, and the cost per bit per second has decreased 20-fold. In the early 1980s, a 1200 b/s modem cost $1000, providing 1.2 b/s per dollar; in 1987 one code obtain 5 b/s per dollar ($2,000 for a 9.6 kb/s V.32 modem); in 1990 one can obtain 16-18 b/s per dollar ($2,000 for a top of the line 38.4 kb/s V.32 modem with error correction, or a 9.6 kb/s V.32 modem for $600). Both the bandwidth and the cost per b/s have improved for the dialup modems as well as for the private line modems. Table 2.2 provides an assessment of both the actual growth of data, and of digital facilities in particular, and may be used to approximate the total data rate in the United States. The long-haul data carried on analog private lines in 1989 is estimated to be 3 billion b/s; that number decreases to 0.3 billion b/s by 1993. (Remember, however, that a lot of data are and will continue to be carried on dial-up facilities). The long-haul data carried on digital private lines in 1989 are estimated to be 23 billion b/s; that number increases to 120 billion b/s by 1993.

Table 2.2 Demands for Telecommunications Services

Transmission Service	Number of Private Lines			
	1989 IXC[f]	1989 LEC[g]	1993 IXC[f]	1993 LEC[g]
Analog dedicated circuit	303000	1035000	36000	100000 [c]
Digital dedicated circuits	47000[a]	88000	310000[b]	600000 [c]
Fractional T1 circuits	1600	(d)	21000	(e)
DS1 circuits	82000	65000	16000	110000 [c]
DS3 circuits	200	2000[c]	1200	10000[c]

a 50% are 56 kb/s lines.

b 90% are 64 kb/s lines.

c Estimated.

d not generally tariffed.

e Unknown if tariffed.

f Interexchange Carrier.

g Local Exchange Carrier.

(i) Compatibility Issues Pertaining to Modems

Given the plethora of techniques available to the modem to encode the digital signal for transmission over an analog circuit, we need to ascertain that the modems at both ends of the circuit are compatible, preferably following some established standard. In theory, two modems designed to the same standard should interoperate. In practice, people generally bay the two modems needed on a link from the same vender; this is particularly true for higher speed modems that maybe using proprietary encoding, modulation, error correction, and data compression schemes.

CCITT has published a series of recommendations to bring some standardization to the equipment. For example, CCITT Recommendation V.22 describes standardized 1200 b/s full-duplex modems for use on the general switched telephone network. V.29 describes standardized 9600 b/s modems for use on point-to-point leased circuits. Table 2.3 depicts some key modem families.

In 1982 CCITT started to develop recommendations for full-duplex two-wire modems operating at 9600 b/s over the public switched network. This was the first time that a modem standard was developed prior to a commercial product; in the past, all modem recommendations had been developed based on successful commercial products. An eight-dimensional error-correcting code is included in recommendation V.32 (two-wire switched line, 9600 b/s) and in V.33 (4-wire dedicated line, 14,400 b/s). V.32 bits (possibly a standard by the end of 1991) is designed to let a V.32-standard 9600 b/s modem operate at 14.4, 12.0, 9.6, 7.2, and 4.8 kb/s in full duplex over a dialup link, with on-line rate negotiation.

The new V.42 standard, primarily concerned with error correction, was formally adopted by CCITT late in 1988. The protocol includes both Microcosm's Networking Protocol (MNP) and CCITT's LAPM (link access procedure for modems). LAPM uses cyclic redundancy checking CRC) to detect errors in transmission and recover with a retransmission. Because an installed base of a half million error-correcting modems using MNP already exists, V.42 included both options. MNP is a de facto standard for error correction that has evolved through several "classes". The basic classes provide error detection and correction; the more advanced classes include compression techniques. Microcom released the first four classes of MNP to the industry and they license Class 5 and Class 6 to vendors that desire to include the protocol in their own products (the higher classes were still proprietary in 1990). Currently, only a couple of vendors have a V.42 product.

Table 2.3 Some Key Families of Modems

Modem Type	Special and Characteristic	1990 Price
Bell 103A	300 b/s full-duplex operation on dial-up line	$20-50
Bell 212A	1200 b/s full-duplex operation on dial-up line	$50-100
V.22 bits	2400, 1200 b/s full-duplex operation on dial-up line	$100
V.22 bits	19,200 b/s full-duplex operation on dial-up line, with compression	$700
V.29	9600, 4800 b/s half-duplex (2-wire) and full-duplex (4-wire) leased line	$800
V.32	9600, 4800 b/s full-duplex operation dial-up line with echo cancellation	$600
V.32	9.6 kb/s to 28.8 kb/s to full-duplex operation on dial-up line with MNP error correction	$1200-2000
V.42	38.4 kb/s-19.2 kb/s with error correction and compression	$1500

V.42 bits aims at providing data compression. Compression ratios of 2-to-1, or even 4-to-1 may be possible. This means that a 9.6 kb/s modem can provide an effective throughput of 38.4 kb/s. the compression is based on an algorithm known as Lempel-Ziv. As of early 1990, only one modem on the market in the United States utilized this technique (the Telebit Trailblazer ™ modem, which operates at 19.2 kb/s, first used an earlier version of a compression algorithm, but now is fully compliant with V.42 bits). Formal approval of the V.42 bits was planned for 1990.

The multitude of modem standards often leads to interoperability problems. Many modem manufacturers now include multiple transmission schemes into a single modem. Such modems allow users to choose an appropriate scheme for the application at hand, and facilitate interoperation with various existing modems. As of early 1990, more than half a dozen companies provided these "multimodems". One such product can, for example, implement nine distinct modulation techniques.

(j) Capacity of a Data Link

The maximum digital capacity of an analog communication channel, in b/s, is given by Shannon's equation:

$$C = W \log_2 (1 + S/N)$$

Where

C = channel capacity in b/s,

W = channel bandwidth in hertz,

S = signal power in watts,

N = noise power in watts.

[If the S/N is expressed as x decibels, then S/N = $10^{(x/10)}$].

Application of Shannon's equation to a voice-grade line with a S/N of 30 dB (which is typical given the amount of power that can be applied to a copper medium, and effects of noise and impairments-39 dB being the theoretical maximum) leads to a maximum bandwidth of 30,000 b/s for a 3000-Hz telephonic voice-grade channel [obtained as 3000 $\log_2 (1+10^3)$]. When a modem has effective throughput of 38.4 kb/s, it is achieved with data compression techniques along with efficient modulation schemes (producing perhaps an uncompressed throughput of 19,200 b/s; the 2-to-1 compression algorithm would produce an effective rate of 38.4 kb/s). This does not contradict Shannon's equation. A higher data rate could also be carried over an unloaded copper wire with no filters because the bandwidth W is higher. A 22-gauge copper wire can, in theory, carry up to 5 MHz, allowing higher digital throughput, as, for example, in the case of twisted-pair LANs, digital loop carrier systems, and ISDN local loops.

(k) Source of Noise in Data Circuits

Many physical factors inhibit successful data transmission. Some of the factors applicable to a traditional telephone channel are Attenuation, Cross talk, Thermal noise, etc., this discussion should clarify the need for digital networks, fiber-based networks, and networks designed specifically for data communication.

1) Attenuation

Attenuation (A) is loss of energy in the signal as it is transmitted through the medium. It is described in decibels and is defined as:

$$A = -10 \log_{10} P_{in}/P_{out} \ \text{dB}$$

Where P_{in} is the input power at one end of the channel and P_{out} is the output power at the other end of the channel. Without amplification, the power out will be less than the power in. for example, a loss of 50% of the power corresponds to a 3 dB loss. A gain of 3 dB is a doubling of power. The measure "dBW" is often used to indicate the gain relative to 1 W of base power. Attenuation affects all types of transmission systems including coaxial-based systems, microwave systems, and fiber optic systems.

In a band-limited channel, such as a traditional long-distance voice circuit, designers customarily design for minimum attenuation in the middle of the spectrum. The loss is usually measured with reference to the 1000-Hz level. The useful bandwidth of a copper circuit is expressed as the frequency difference between points on the attenuation curve that represent 10 dB of loss with respect to the level of a 1000-Hz reference signal.

Attenuation versus signal level refers to dynamic loss in the channel, such as that caused by a compander operating at a syllabic rate. A compander is a device installed in the voice channel that amplifies weak signals more than strong signals at the source, and amplifies strong signals more than weak signals at the destination, to restore the original characteristics. It is fast acting, following the syllables of speech for maximum effect. The companding technique permits the voice signal to be transmitted at a higher average level than would otherwise be the case and results in a consequent improvement of the S/N at the receiver. In data transmission, companding has the undesirable effects of decreasing the apparent modulation percentage while producing spurious frequencies that show up as noise. Line conditioning can ameliorate the situation.

2) Cross talk

Inductive, capacitive, or conductive coupling between adjacent circuits results in having an undesired signal "leak" across and appear as a weak background signal added to the desired signal. Cross talk degrades the S/N and increases the error rate. A limit is placed on the amount of noise that is allowed on the circuit. If the noise can not be attenuated nor than 24 dB below the signal, the circuit may need to be repaired or changed.

3) Echo

When an electrical circuit such as a telephone line is not properly terminated, some of the data signal energy is absorbed by the receiver and some energy, called echo, is returned to the sender. A portion of the echo may be reflected by an improper line termination at the sender and reach the receiver as a noise. Like other kinds of noise, it increases the error rate, unless the signal level of the data is much greater than the signal level of the echo. The ability of a circuit to reduce reflections is called echo return loss and is measured in dB. The greater the return loss, the better is the termination. A ratio of 24 dB or more is desired.

4) Transients

Transients include impulse noise, gain hits, phase hits, and dropouts. Impulse noise is defined as any noise that exceeds the root mean squared level of the background noise by 12 dB for more than 10 ms. This kind of noise can be expected on most metallic facilities. It is the result of coupling from nearby circuits that carry electrical current surges originating in switching equipment, line faults, or from lightning surges. The desired data signal usually exceeds the background noise by at least 26 dB. A correlation exists between the frequency of occurrence of noise impulses and the number of electromechanical switching offices through which the data circuit must pass. Reduction of these switches in the 1980s has had the effect of decreasing the importance of switches as a source impulse noise.

Telephone circuits are rated according to the average number of "hits" of impulse noise that exceed threshold over a period of time. When the statistical average is exceeded, an attempt ought to be made to find the cause and to make the necessary repairs. A typical rating is 15 counts in 15 minutes. Impulse noise hits tend to be clustered with long periods between clusters; also, a typical noise impulse is much larger than the data signal, and mutilates several successive data bits. Impulse noise affects any kind of modem and any modulation scheme. It will affect more data in a high data rate stream then in a slow one because of the disruptive effect of a larger hit on the modem. The terminal equipment user should be prepared to cope with various patterns of impulse noise by providing data streams that can be acknowledged within the average interval of noise hits. If data blocks are too long, the probability is high that each one will receive a noise hit and require retransmission (with the chance of another hit. On the other hand, if blocks are short, the overhead incurred by the synchronizing sequence, header, and acknowledgments will reduce transmission efficiency.

5) Thermal Noise

Thermal noise (also called broadband or white noise) arises from the thermal agitation of electrons in resistors and in semiconductors. The greater the absolute temperature of the noise source, the greater the noise level. In telephone systems, this type of noise appears as a background hiss. Its strong appearance usually indicates a faulty component. The usual effect of thermal noise is to cause infrequent and random single-bit or double-bit errors, until a modem signal-to-noise threshold is reached, after which the error rate becomes catastrophic.

6) Intermodulation Distortion

Nonlinear components in the network will produce distortion of the data signal. The distortion becomes harmful if the time constant of the nonlinearity is short compared to the frequencies that compose the data signal. Intermodulation results in the production of spurious frequencies by a nonlinear medium. These frequencies are the sum and difference of fundamental frequencies present in the undistorted analog data signal, as well as the sum and difference between their harmonic frequencies, and between fundamentals and harmonics. The level of intermodulation distortion is typically less than 5% of the desired signal level and appears as correlated background noise that can cause characteristic distortion of the data signal (i.e., weak bit patterns).

7) Delay Distortion

In the process of transmission, all frequencies may not arrive at their destinations with the same relative phase relationships that existed at the transmission point. The effect is that some portions of the data signal appear to be delayed in time, with reference to other portions of the data signal. A modulated signal has a fundamental frequency plus harmonic frequencies. Each harmonic wave has a fixed phase relationship with the fundamental wave that must be preserved if the data signals are to be reproduced by the receiver. When the harmonics are advanced or retarded in phase, they appear to arrive early or late. Some signal components overlap the time domains of adjacent signal elements, causing intersymbol interference. The solution to the delay distortion problem is called delay distortion.

8) Incidental FM

This phenomenon is peculiar to frequency-based carrier equipment, for example, in a link with analog microwave equipment. Incidental FM usually arises from the influence of varying power supply voltages on the oscillators used in the equipment. If the oscillators are not stabilized, hum frequencies and lower frequency ringing voltages can modulate the oscillators, both in frequency (or phase) and in amplitude. The amplitude effect is usually of lesser consequence. The effect of phase jitter on a data signal is to decrease the certainty of detection of data bits in the receiving modem. High-speed modems that use PSK are most vulnerable. The pool of analog carrier systems is becoming smaller in the United States, as fiber is increasingly deployed, diminishing the importance of this impairment.

9) Frequency Error

This phenomenon, like incidental FM, is endogenous to carrier frequency equipment. It results from the fact that the heterodyning oscillators in location A can differ in the frequency by several hertz from those in location B, with the result that the data carrier and wideband frequencies are offset. This is not noticeable to a telephone listener, but modem performance can be degraded slightly, and a poorly designed modem will fail to receive data correctly.

(I) Line Conditioning

Tow option are available to reduce the effects of attenuation and delay distortion in a traditional data circuit: conditioning and equalization. Line conditioning may be used when data transmission on dedicated voice-grade lines occurs at speeds of more than 4800 b/s. line conditioning will ensure that the circuit conforms to transmission specifications and is less susceptible to errors. Line conditioning is provided on leased lines by carriers at a monthly fee. The carrier may add special equipment to the circuit; conditioning provides a method to diminish the problems of attenuation and delay, but it dose not totally eliminate the impairments; it provides for more consistency across the bandwidth. For attenuation, the carrier adds equipment that attenuates the frequencies in the signal that tend to remain at a higher level than others. Thus, attenuation still occurs but is more evenly distributed across the channel. Conditioning is not available over the switched telephone network.

Two types of line conditioning are available in the United States: C-conditioning and D-conditioning. C-conditioning is used to deal with attenuation distortion, which occurs when the relative amplitudes of the different frequency components change during transmission. This can occur because of uneven attenuation or uneven amplification. D-conditioning is used to deal with harmonic distortion, which is caused by the presence of unwanted harmonics from the input signal in the output signal. D-conditioning also improves the S/N. the standard specification requires a S/N of not less than 24 dB, a second harmonic distortion of not more than 25 dB, and a third harmonic distortion of not more than 30 dB. With D-conditioning, the carrier gives a S/N ration specification of 28 dB, a signal to second harmonic distortion ration of 35 dB, and a signal to third harmonic distortion of 40 dB. D-conditioning is accomplished by eliminating or avoiding noisy facilities for the given circuit.

An attenuation equalizer adds loss to the power frequencies of the signal because these frequencies decay less than the higher frequencies in the band. The signal loss is then consistent throughout the transmitted signal. After equalization is applied, amplifiers restore the signal back to its original level. A delay equalizer compensates for total signal delay. The higher frequencies thus may reach the receiver ahead of the lower frequencies. Consequently, the equalizer introduces more delay to these frequencies to make the entire signal propagate into the receiver at the same time.

With the recent development of sophisticated modems able to cope with problems on dialup lines (and thus, by extension, in private lines), and with the continued introduction of digital lines, the importance of conditioning, as currently defined, may diminish in the 1990s.

(m) Data Error Detection and Correction

Data communication systems require error control mechanisms to deal more explicitly with the problems listed in the previous section than with line conditioning. In a metropolitan environment with considerable man-made electromagnetic noise and multipath problems, the need for error correction is imperative, especially for data carried on copper loops or radio systems.

The original work on error-control coding was undertaken in the 1940s and 1950s by Shannon, Hamming, and Golay. Coding is now a mature branch of communication, with a strong mathematical foundation. These advances in theory combined with the emergence of inexpensive VLSI provide cost-effective means to achieve efficient and highly reliable communication.

In terrestrial links (where the signal propagation delay is small), the commonly used method of error control is the automatic repeat request (ARQ). Once an error is detected, the receiving system asks for retransmission from the sender (automatic refers to the fact that no user intervention occurs). Two types of ARQ systems exist: (a) stop-and-wait ARQ and (b) continuous ARQ. In the stop-and-wait ARQ, the sender system waits for an acknowledgment from the receiver system on the status of the transmitted block. If the acknowledgment is positive, than the sender system transmits the next block; if it is negative, it repeats the transmission of the blocks that have errors. This approach unsuitable in a satellite environment where a round-trip delay of half a second is required for the reception of the acknowledgment. A more acceptable form of ARQ is the continuous ARQ. With this approach the sender sends the blocks and receives the acknowledgment continuously. Once a negative acknowledgment is received, the transmitter sends either the block with an error and all blocks that follow it, or sends only the block that has the error.

Both of these ARQ approaches require a duplex channel, so that handshake information can be sent back and forth. If the channel is simplex, as is the case with the FM-sub carrier (or the propagation time is so long that the channel is effectively simplex-for example, communicating with an interplanetary spaceship), then a totally different method is required. The form of error control commonly used in satellite systems is the forward error correction (FEC). In this system, extra bits of data are added to the blocks for error checking and correction. If an error is detected, enough redundant information is carried along, which permits the receiving end to fix the incoming message; the receiver need not go back to the sending party to obtain a retransmission as in ARQ.

In FEC, one wants to be able to remove all redundancy from the source of the information, so that the amount of data to be transmitted is minimized. The channel encoder performs all the digital operations needed to prepare the source data for modulation. The encoder accepts information at rate R_s and adds its own redundancy, producing an encoded data stream at a higher rate R_c. There are two types of FEC methods, the block codes and the convolutional codes.

When using a block code, the encoder accepts information in sequential k bit blocks and, for each k bit, generates a block of n bits, with $n \geq k$; the n-bit block is called a code block or code word. The ratio k/n is called the rate of the code. Thus, the stream of data is broken into k symbols of information and n – k redundant symbols for error control, where n is the total length of the code word. The resultant system is referred to as (n,k) block code, of which many forms exist; the most popular ones are the cyclic codes, which can simply be formed from cyclic shifting of bits of data in a block, thus creating a code word. The two most common coding techniques are the BCH (Bose-Chaudhuri-Hocquenghem) code and the Golay code.

Convolutional codes are another class of FEC methods. For encoding with a convolutional code, the encoder accepts information bits as a continuous stream and generates a continuous stream at a higher rate. The information stream is fed to the encoder b bits at the time (b typically ranges from 1 to 6). The encoder operates on the current b-bits and some number k (called constraint length) of immediately preceding b-bits inputs to produce B output bits, with B > b. Here the code rate is b/B. the encoder for the convolutional code might be thought as a form of digital filter with memory extending k – 1 symbols in the past. A typical binary convolutional code will have b = 1, B = 2 or 3, k = 4 or 5 or 6 or 7 (in special situations k can be as high as 70). The channel decoder undertakes the conversion of the demodulator output into symbol decisions, which reproduce as accurately as possible the data that were encoded by the channel encoder. The most widely used convolutional code is the Viterbi code. Viterbi coders that transmit 256 kb/s are now available for less than $90 in quantities.

Trellis code modulation (TCM) is a relatively new technique now available in high-speed modem. The method is such that the signal (derived and coded from the user data bit stream) is allowed to assume only certain characteristics (states). User bits are interpreted such that only certain of the states are allowed to exist from prior states. The transmitting device accepts a series of user bits and develops additional (restricted) bit patterns from these bits. Moreover, a previous user bit pattern (a state) is used to determine the current bit patterns (states). Certain other states are not allowed and are never transmitted. The transmitter and receiver are programmed to understand allowable states and the permissible states transitions. If the receiver states and state transitions differ from redefined conventions, we assume that en error has occurred in the circuit. By convention, the transmitter and the receiver know the transmission states and permissible state transitions; the receiver analyzes the receiver signal and makes a "best guess" as to what state signal should assume. It analyzes current states, compares them to previous states, and makes decision as to the most relevant state. If effect, the receiver uses a path history to reconstruct damaged bits. Trellis coding is an error-correction code with a memory. It increases the BER performance on a line by two to three orders of magnitude.

(n) Parity Checking

Parity checking is a simple but relatively unreliable method for basic error detection. It was primarily used in the late 1960s and early 1970s. Parity check schemes are simple examples of block codes. Here the encoder accepts k information bits and appends a set of r parity check bits, derived from the information bits according to sum predefined algorithm. The information and parity bits are transmitted as a block of n = k + r bits. A typical parity code is (8,7). Single parity check codes lack sufficient power to provide reliable communication. Hamming codes provides more powerful block codes and are also used for error control in computer memorize and other mass storage systems.

Two versions exist: an odd-parity and an even-parity. In even-parity, the number 1s in the 7-bit ASCII representation of each of each character being sent is counted. If that number is even, a 0-bit is concatenated to the 7-bit code; if the number is odd, a 1-bit is concatenated to the 7-bit code, thus giving an even number of bits (odd-parity does the reveres). For example, the code 1111111 becomes 11111111. The receiving end will count the number of ones; if there are an odd number of ones, the end system concludes that an error must have occurred in transmission.

The problem with this method is that line hits and dropouts tend to occur in bursts, affecting several contiguous bits, particularly at high speed. If more than one bit is affected, the method may fail to detect an error. For example, 11111111 becomes 00000011, the receiving end will not be able to detect the problem.

(o) Cyclic Redundancy Checking

Cyclic redundancy checking is the prevailing method used in conjunction with ARQ to detect errors in long blocks of data. The process of generating a CRC for a message involves dividing the message bye a polynomial, producing a quotient and a remainder. The remainder, which usually is two characters (16 bits) in length, is appended to the message and transmitted. The receiver performs the same operation on the received messages and compares its calculated remainder. If the two CRCs fail to the match, the protocol causes the block to be to be discarded, and a retransmission is requested. In contrast with the parity method discussed above, and other methods used in the 1970s (vertical and horizontal error checking), the undetectable error rate for CRC-protected data is extremely small (the undetectable error rate depends on the length of the CRC and the length of the data block).

2.3.2.3 Switching

Switching is as important for data communication as it is for voice communication. Six types of switching applicable (but not exclusively) to data are described below.

(a) Circuit Switching

In a circuit switched connection, the end-to-end path of a fixed bandwidth exists only for the duration of the session. The destination is identified by an address; the network receives the address from the sender and sets up a path, typically within seconds, to the destination. At the completion of the call, the path is taken down. Circuit switching is not only suitable for voice transmission, which employs this method almost exclusively, but also for data transmission.

Circuit switched connections are economical if the end-to-end session is short (on the order of minutes), or the geographic area to be covered is wide. However, if the sessions are long and traffic to a specific destination is heavy, the cost of using circuit switching may exceed the cost incurred by other methods (for example, a dedicated line). If calls are very short, the setup and teardown overheads may be excessive; other methods (such as packet switching discussed below) may be better.

(b) Channel Switching

Channel switching refers to a service that allows the user to establish a channel that can stay in place for hours or days, and can be reterminated (typically in minutes) as needed. The nomenclature used to describe this service varies, and the terms "reserved," "semipermanent," and "permanent" have all been used. Channel switching can be considered an extension of circuit switching with the following four variations: (1) the call setup time is in minutes rather than in seconds; (2) the duration of the call is in hours or days rather than in minutes; (3) it employs "slow" switches in the CO, such as a digital cross-connect system, rather than a traditional circuit switch; and (4) it is cheaper for the user, compared to straight circuit switching, for long session. Channel switching may be provided under ISDN, although manifestations appeared in the early 1980s using digital cross-connection systems. It is a service positioned between circuit switching and dedicated lines (no switching).

(c) No Switching

Many data applications use dedicated lines (also known as a "private line") that do not include carrier-provided switching. The line is leased at a fixed monthly rate, and the charge is independent of usage. Once installed, these facilities can stay in place for years. T1 circuits now commonly used as backbone data networks, are examples of dedicated channels. Systems Network Architecture (SNA), predominantly employs dedicated lines, although a packet switched interface is also available.

(d) Packet Switching

Early approaches to data communication were based on techniques for voice communication such as circuit switching. One soon discovered that the dynamic allocation of bandwidth would allow more efficient utilization of available network resources for interactive data communication. Packet switching has emerged as an important approach in data network. In packet switching, information is exchanged as block of limited size or packets. At the source, long messages are reassembled at the destination to reconstitute the original message. Many users can share network resources, although efficient use of transmission resources increases the network complexity. Data buffers are needed at each node; however, the storing is typically transient in nature, and should be of the order of tens or hundreds of milliseconds. Packet switching can be viewed as a case of message switching, except that the very formal procedures are used (absent in traditional message switching), and the period of nodal storage is small, as indicated.

Packet switching comes in two types: connection-oriented (such as in traditional x.25 wide-area networks), and connectionless (such as in LANs and MANs).

(e) Message Switching

Message switching refers to a method of storing a message at intermediary nodes in the network for nontrivial amounts of time (i.e., more than a couple of minutes). This method was commonly used in telegraphy and telex networks. It almost disappeared, but may now be reemerging in store-and-forward E-mail, particularly with the message handling system.

(f) Distributed Switching

LANs, MANs, and IBM's SNA, among other systems, employ a form of a distribution "self-switching."* In this environment, the data are labeled with an address and are broadcast to every user connected to the particular transmission system at hand (for example, in a LAN this would be every user on the bus or ring). Each user's equipment receiving the message examines the addressed on it and is trusted not to display or throw away the data unless the message is addressed to that user. This technique is become prevalent, and it has the advantage of being efficient because it relies on distributed user-provided intelligence to carry out the message sorting task; the sorting and switching task would have to be done centrally, by carrier-provided resources. Additionally, it can be more reliable, if properly designed, because failures of a single component dose not necessarily affect the entire communication system.

2.3.2.4 Layered Protocols

(a) Motivation

As we discussed at the beginning of this chapter, communication involves by definition two (remote) entities, also called end systems. The tow entities should be peers, namely, enjoying the same set of communication privileges, although this is a relatively new approach (throughout the 1960s and 1970s, a master-slave approach was much more typical). To undertake communication, a fairly large number of functions must be carried out. In addition, agreements must exist between the two end systems on how to undertake functions that have remote importance. These agreements are now known as protocols, and publicly agreed protocols are referred to as standards.

A layer is a defined set of related communication functions. Protocols describe ways in which remote peers can utilize functions within a layer. Layering (or modularization) provides the following benefits, among others:

1. Easier understanding of the communication process by working with a small number of logical groupings;
2. Collecting related functions in the same groupings minimizes the number of interactions between layers and simplifies the interfaces;
3. Layers can be implemented differently and changed to take advantage of new developments without affecting the other layers;
4. Simple layer boundaries can be created with at most only two neighbors.

   (Layering by itself dose not necessarily imply peer-to-peer communication capabilities: a number of vendor-specific architectures of the 1970s employed the layering concept (i.e., IBM's SNA, DEC's DECNET); however, the open-layered architectures that have evolved in the 1980s can provide peer-to-peer capabilities.)

(b) Open Systems Interconnection Reference Method

To facilities interconnection, standards for open systems have been developed by the International Organization for Standardization. Seven major layers have been defined in what is now known as the Open Systems Interconnection reference model (OSIRM), which has been available since 1984, as follows: application (7), presentation (6), session (5), transport (4), network (3), data link (2), and physical (1) layers. See Table for a listing of some of the key functions of the OSIRM layers. (The application layer should properly have been named "application support layer" because the ultimate user application utilizing communication facilities to interact with partners in other systems resided above the application layer)> this model is described in specification ISO 7498 and also in CCITT X.200. the term upper layers are in the direction of the layer (3) through (1).

All contemporary descriptions of data communication (and telecommunication, for that matter), including network management, security, addressing, and internetworking, employ the frame work defined in the OSIRM. The higher adjacent layer is called the user; the lower one is called the provider describe, respectively, the relationship between the consumer and the producer of a layer service. As one moves through layers, users become providers and vice versa.

Table 2.4 Layers of the Open System Interconnection Reference Model

Layer	Function
Application	Support of user functions such as file transfer, transaction processing
Presentation	Transfer syntaxes (character coding)
Session	Coordination services, dialogue, synchronization
Transport	Reliable end-to-end communication
Network	Delivery within a single sub network; end-to-end, such as addressing and internetworking
Data Link	Delivery of blocks of data between two points
Physical	Bit transmission

Entities exist in each layer. An entity is an active element within a layer that carries out the layer's prescribed functions. A layer may contain multiple entities for different functions. Entities in the same layer, but in different systems, that must exchange information to achieve a common objective are called "peer" entities. Peer defines an equivalent layer entity in another system within the open environment. A peer may be of a different hardware or software environment, but behaves consistently in all cases. Entities in adjacent layers interact through their common boundary.

The application layer can be viewed as an extension of local operating system supervisory functions to another system. These supervisory functions include the following: (1) identifying the intended partner and activating authentication procedures; (2) agreeing on quality of service, security, payment, etc.; (3) supplying services to control the modification of shared data; (4) determining that required resources are available; and (5) specifying more detailed application requirements.

The upper layers – application, presentation, session, and transport – are generally, although not always, independent of the telecommunication network; the reason for the exception is that some carriers may offer functionality above the network layer, for example, E-mail. In general, however, these layers are components of the end-user systems and are insulated from networking operations. The interworking layers create computer-system-computer services operating across any combination of subnetworks. The OSIRM layering is applicable both when communicating over a carrier's network and when communicating over a private (CPE) network.

Communication with a remote peer, at the same layer, involves a protocol. Adjacent entities communicate by exchanging primitives with each other via the service access point (SAP). The SAP is a conceptual delivery point, and as such it can be addressed. Note, however, that the relationship between SAPs and entities is not one to one.

(c) (N)-Layer Formalism

Some formalism related to important concepts associated with a layer, already described above in more intuitive form. This formalism will be required when discussing name and addressing issues. The functionality of a generic (N)-layer is provided by one or more (N)-entities. These (N)-entities must be identified and located to perform communication. An (N)-title is a name that uniquely identifies a particular (N)-entity throughout the (N)-layer in an OSI environment; an (N)-title is independent of the location of an (N)-entity. (N)-service access points represent the logical interfaces between (N)-entities and (N+1)-entities. An (N)-SAP address is a name that identifies a set of one or more (N)-SAPs to which an (N+1)-entity is attached. An (N)-SAP address relates to an (N)-SAP, and not directly to an (N+1)-entity. Hence, an (N)-entity is identified by an (N)-title, independently of the (one or more) (N – 1)-SAPs to which the (N)-entity may be bound. An (N)-entity is located by specifying the (N – 1)-SAP address of the (N – 1)-SAPs to which the (N)-entity is bound. A mechanism exists at each layer which associates (N)-entities with their (N – 1)-SAPs; this is the (N)-directory. The (N)-directory maps (N)-title of (N)-entities onto the (N – 1)-SAP address through which they communicate. An OSI address comprises nested (N)-SAP addresses.

(d) Primitives and Services

A primitive is the smallest unit of action that can be specified. More precisely, it is a conceptual instruction in a layer service. The primitives represent, in an abstract way, the logical exchange of information between a layer and the adjacent layers; they do not specify or constrain implementation. Examples of primitives are: establishing communication with a remote peer, sending data, and inserting a synchronization point. Most communication activities require a number of primitives to complete their tasks. A service primitive consists of a name and one or more parameters passed in the direction of the service primitive. The name of the service primitive contains three elements: (1) a type indicating the direction of the service primitive, (2) a name, which specifies the action to be performed, and (3) an initial (or initials), which specifies the layer (or sublayer) providing the service.

Two kinds of the services are available: confirmed and unconfirmed. A confirmed service produces information from the remote peer entity on the outcome of the service request (this may be needed when additional action is contingent on a successful outcome). An unconfirmed service only passes a request along; this is a faster interaction because no overhead is involved with the response.

Four generic types of service primitives for confirmed service are:

1. Request: a service request from a higher layer to a lower layer (more formally: a primitive issued by a service user to invoke a service element);

2. Indication: a notification from a lower layer to a higher layer that a significant event has occurred (more formally: a primitive issued by a service provider to advise that a service element has been invoked by the service user at the peer service access point or by the service provider);
3. Response: the response to a request (more formally: a primitive issued by the service user to complete at a particular service access point, some service element whose invocation has been previously indicated at that service access point);
4. Confirm: message passed from a lower layer to a higher layer to indicate the results of a previous service request (more formally: a primitive issued by a service provider to complete, at a particular service access point, some service element previously invoked by a request at that service access point).

Peers exchange protocol data units (PDUs) containing (1) protocol control information (PCI) and (2) data. A user initiates activity by issuing a service request across the SAP. The entity receives request and constructs a PDU, the type and values of which are determined by the request and locally available information. The PDU is delivered to the remote peer partner using the services of the underlying layers (the PDU will be enclosed as data in a subsequent service request to a lower layer. When the remote entity receives the PDU, it generates a primitive that it passes upward via the SAP to the user.

More details on the OSIRM and related standards are provided; however, a basic understanding is required at this juncture for use in the intervening chapters. ISDN, broadband ISDN, signaling, LANs, MANs, network management, and security all require an understanding of the OSIRM and related recommendation. We describe the first three layers of the OSIRM below. The entire model is described.

(e) Layer 1 – Physical

This layer deals with the physical connection of the circuits. Recommendations at this layer standardize pin connections between DTEs and DCEs and are also concerned with the transmission of bits between machines. Examples of protocol standards for this layer are RS-232-C and RS-449, noted earlier for connection between terminal devices and modems, and I.430 for ISDN.

(f) Layer 2 – Data Link

This layer deals with data transmission over a signal link. The protocol in this layer detects and corrects in the transmission. This layer creates frames for the data to be transmitted, including a CRC code. The functionality includes sending acknowledgments to the sender end system. It will signal for retransmission if a frame is out of sequence or is mutilated. This layer uses flags and headers in the frame, so that the receiver and system can recognize the start and end of a frame. Typical standards for this layer are: IBMs Synchronous Data Link Control (SDLC), ISO's High-level Data Link Control (HDLC), and ISO's Link Access Procedure B (LAP-B).

(g) Layer 3 – Network

This layer receives message from the higher layer, segments them into packets, and sends them to the receiver through the data link physical layer. At the receiving end, functionality in this layer reassembles the message into original form. It enables routing, multiplexing, and flow control. Typical standards for this layer are: CCITT's X.25 Packet Level Protocol and ISO's 8473, which deals with internetworking.

2.4 Telecommunication Carriers

This section describes the commercial providers of telecommunication services. In the United States, a common carrier is an agency that provides telecommunication services to the general public. A common carrier typically provides a spectrum of services. Descriptions of services and rates are called tariffs, and are filed with the Federal Communications Commission (FCC) or with the local state public utility commissions (PUCs). Common carriers can be grouped as follows:

1. Exchange-access carriers (intra-LATA), such as the Bell Operating Companies (BOCs) and independent telephone companies; these are collectively known as LECs.
2. Interexchange carriers (IXCs), such as AT&T, US Sprint, and MCI; and

3. Specialized common carriers (SCCs), such as Telenet; SCCs are like other common carriers, but specialize in providing specific services (for example, data or video communication); these are also called value-added carriers, or value-added networks (VANs).

While 1500 or so local telephone companies provide exchange access common carrier services, 85% of the business is provided by the BOCs. Approximately 5% of the LEC's customers account for 50% of the revenues. Approximately 200 IXCs exist in the United States.

Other types of carriers also exist. A private carrier offers point-to-point links, has no switched service, negotiates with customers on an individual basis, and has a majority of leases that are long-term, with a limited and stable customer base. A company that offers only nonswitched point-to-point service to business customers, and dose not offer any services to the general public, can be considered a private carrier. These carriers are also referred to as "noncommon carriers." A nondominant fiber optic carrier is a carrier that provides principally or solely end-to-end fiber-based communication links to another carrier or to end-users, generally in an inter-LATA arrangement. Teleport is, formally, an access facility to a satellite or other long-haul telecommunication medium that incorporates a distribution network (usually fiber-based) serving the greater or other economic development. In practice, however, most teleports are not usually involved in major real estate or local economic development. About 20 teleports operate or plan to operate in the United States.

2.5 Network Design Philosophies

In reading this material, keep in mind that, depending on the objective, one may be led to design and use different network topologies, technologies, and architectures. For example, some objectives could be to:

- Minimize cost (build the cheapest network);
- Maximize the cost-performance ratio (get the most network for the dollar);
- Maximize profit (build a network that allows the firm to be very aggressive, reaching new markets, et cetera);
- Maximize profit rate (especially for a utility);

- Minimize risk of loss (military network);
- Maximize safety (network designed for police or fire departments);
- Maximize quality of services (for a given investment);
- Maximize growth opportunity for the firm (build a network that can easily grow in the future – this may be the course of action of a carrier);
- Maximize prestige of the firm (i.e., buy the newest equipment to impress investors, competitors, public);
- Many other possible objectives.

Clearly, these criteria are not all compatible with each other: a network built to minimize the cost will probably not maximize reliability and quality of service. Hence, the telecommunication manager must bee familiar with the dynamics of the design synthesis. Complexity in design (and corresponding responsibility for the planners) continues to increase inexorably. Using 1985 as an arbitrary reference for discussion and ignoring the effects of divestiture, the planner must now deal with the following new technical issues and options (some of which were only embryonic at that time):

- Availability of high-speed dialup modems operating up to 38.4 kb/s, and cellular telephony modems operating up to 16.8 kb/s, which make the optimization process compared to dedicated lines and ISDN an interesting one;
- Proliferation of personal computers and workstations in the office and on the factory floor, and their need to communicate, probably through internet worked LANs. High-speed backbone networks operating at 16 and 100 Mb/s are also becoming available. In addition to coaxial-based solutions, the planner can choose unshielded twisted-pair and multimode or single-mode fiber;
- Fourth-generation PBXs that not only allow integration of data, but also include LANs, resource servers (file servers, communication servers, protocol gateways, etc.), ISDN access, and advanced internal algorithms such as "dynamic bandwidth allocation." Also, we have witnessed the substantial improvement of CENTREX services, making it more

competitive with PBXs, and rendering the decision-making process in choosing a solution more difficult.

- Availability (in the United States) of high-speed digital interfaces to the telephone plant, to another networked PBX, or to a mainframe computer via a CPE T1 mutiplexer. T1 multiplexers are being introduced in private networks at an ever-increasing for the purpose of corporate-wide voice, data, and video integration. Fractional T1service is also becoming important, and there is talk of fractional T3;
- The introduction of high-speed channel-to-channel communication among mainframe computers, and between mainframe peripherals;
- The ongoing introduction of ISDN with its user-to-user and user-to-network signaling capability, and advanced services, including network-based automatic call distributor (ACD), and automatic number identification (ANI), which facilitates integrated voice and data applications. Switched Multi-megabit Data Service (SMDS), with data rates in the 1.5 to 45 Mb/s range, and broadband ISDN (BISDN), with data rate in the 150 to 600 Mb/s range, are also on the horizon;
- The possibility of interconnecting dispersed LANs via MANs and potentially and potentially using fiber optics facilities;
- The emergence of very small aperture terminal (VSAT) satellite networks, allowing cost-effective two-way communication over four-foot dishes, and other radio services for data usage;
- The increased need for network management: networks can cost millions of dollars every year; optimal usage is critical;
- Security has become a major area of concern as more and more computers are connected to networks, making them accessible and susceptible to attacks.

All of these factors and technologies complicate the design process; understanding the new design requirements and acquiring the appropriate tools is a mandatory effort in establishing cost-effective, reliable, and flexible cooperate networks. The responsibilities of a data communication manager are becoming more demanding as industry competition increases along with the ensuing multiplicity of available communication options and the increased obsolescence rate of this technology. In designing a network, multiple options must be analyzed and the issue of hidden costs associated with these options must be taken into account. This richness of communication options offers the data communication manager major opportunities, not only in terms of optimal network design, but also in terms of network management, grade of service, and reliability. However, analytical methods are required to facilitate the decision-making.

The multiplicity of options makes the task more difficult in the decade of the 1990s. Additionally, deployment of the least expensive network may not always be the best course of action if the network constricts the company's ability to be competitive, deliver products and services quickly, grow smoothly. As we can see, the problem is not a trivial one. Due to the high multidimensionality of the solution space.

CHAPTER 3

FUTURE DEVELOPMENTS IN TELECOMMUNICATIONS

3.1 Introduction

The 1980s and 1990s was marked by rapid development of telecommunications services and technologies. We currently use services daily that were not available ten to fifteen years ago, such as LANs, cellular phones, and graphical Internet. It is expected that this technological development and the growth of telecommunication business will continue for several years. Examples of such expected phenomena as well as the technologies and services under development that will be put into use during the coming few years are:

- High penetration of low-cost mobile services, PCS;
- Competition in local networks through the WLL technologies;
- Digital high-quality and high-capacity broadcasting systems;
- Introduction of new services using the intelligent network concept;
- Extremely high-capacity optical networks (terabits/s);
- Multimedia communications with improved quality;
- Customized media presentation;
- Interactive video services;
- Safe and user-friendly electronic shopping services;
- Speech recognition and synthesis systems;
- Pocket size "smart" mobile terminals;
- Real-time language translation;
- Video-on-Demand services;
- Electronic libraries;
- Integration of telephone service and Internet.

It is difficult to estimate which new services will get market acceptance and which will not. A technology must be available; but, in addition, success depends on many other things such as how the new services are launched and charged, what alternative services are available, and timing of the launch. In the following sections we will look at some new technologies and services in more detail.

3.2 Optical Fiber Systems

The major transmission networks will use optical fiber transmission systems that provide much higher capacity than the systems of today. The optical fibers will one day be used in the LANs just as they re used in trunk networks today.

WDM provides a higher capacity than fiber cables. Through WDM, many systems can use the same fiber if they operate at a different wavelength. At the receiving end optical signals are separated from other signals with optical fibers.

In local networks the technology known as passive optical networks (PON) may be put into use. In this technology, optical couplers are used to split and combine optical signals to and from subscribers, which is assumed to make the use of optical transmission in local networks economically feasible.

Coherent optical systems will increase the capacity of fibers dramatically. Present optical systems merely send light pulses and use the whole bandwidth of the fiber for a single signal. Coherent technology means that we use light as a carrier in the same way as we use radio frequencies in present radio systems. Then we could modulate multiple carriers to the fiber and use the whole bandwidth of the fiber efficiently.

3.3 Mobile Communications

The capacity of the cellular mobile systems will be increased with sophisticated speech processing technology, microcell and picocell structure of the networks, and new frequency bands that are put into use. Multimode mobile terminals that are able to access different networks are becoming available. These terminals may operate like a cordless telephone in an office environment, cellular telephones at different frequencies (GSM, DCS1800, PCS1900, and CDMA), and even as satellite mobile telephones.

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TABLE OF CONTENTS

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I am and shall always be in your gratitude. With all my love, respect and best wishes, thank you all so much.

ABBREVIATIONS

PTT	Post, Telegraph and Telephone.
PSTN	Public witching Telephone Network.
PLMN	Public Land Mobile Network.
MAN	Metropolitan Area Network.
WAN	Wide Area Network.
LAN	Local Area Network.
ATM	Asynchronous Transfer Mode.
ISDN	Integral Service Digital Network.
SONET	Synchronous Optical Network.
SMDS	Switched Multi-megabyte Data Service.
FDDI	Fiber Distributed Data Network.
CO	Central Office.
AT&T	American Telephone & Telegraph.
RBOC	Regional Bell Operating Company.
NCR	National Cash Register.
GTE	General Telephone and Electric.
CATV	Common Antenna Television.
PCS	Personal Communications Services.
MFS	Macintosh File System.
MCI	Media Control Interface.
FCC	Federal Communication Commission.
IXC	Inter Exchange Carrier.
SNET	Southern New England Telephone.
TCI	Tele-Communications Inc.
CLEC	Competitive Local Exchange Carrier.
DSL	Digital Subscriber Line.
TELCOS	Name of a Telephone Company.
OSS	Operation Support System.
COE	Central Office Equipment.
DMS	Digital Multiplex System.
FMT	Fiber Multiplex Transport.
DACS	Digital Access and Cross-Connect System.
RCU	Remote Carrier Urban.
RSC	Remote Switch Concentrator.
RLCM	Remote Line Concentrating Module.
FMS	Forms Management System.
MFA	Mechanized Frame Administration.
IP	Internet Protocol.
SLAT	System Line Up and Test.
WDM	Wavelength Division Multiplexing.
PON	Passive Optical Network.
GSM	Global System Mobile.
CDMA	Code Division Multiple Access.
N-CDMA	Narrowband-CDMA.

UMTS	Universal Mobile Telecommunication service.
TDMA	Time Division Multiple Access.
DECT	Digital European Cordless Telecommunications.
PAC	Privilege Attribute Certificate.
SDH	Synchronous Digital Hierarchy.
WLL	Wireless Local Loop.
RLL	Radio Local Loop.
VoD	Video on Demand.
VPN	Virtual Private Network.
CTI	Computer Telephone Integration.
VOIP	Voice Over Internet Protocols.
CBR	Constant Bit Rate.
PCM	Pulse Code Modulation.
B-ISDN	Broadband-ISDN.
HDTV	High Definition Television.
DPCM	Deferential PCM.
ASCII	American Standard Code for Information Interchange.
ASK	Amplitude Shift Keying.
FSK	Frequency Shift Keying.
PSK	Phase Shift Keying.
DTE	Data Terminal Equipment.
DM	Delta Modulation.
FM	Frequency Modulation.
AM	Amplitude Modulation.
PM	Phase Modulation.
PAM	Phase Amplitude Modulation.
QAM	Quadrature Amplitude Modulation.
BER	Bit Error Rate.
MNP	Microcosm's Networking Protocol.
LAPM	Link Access Procedure for Modems.
CRC	Cyclic Redundancy Checking.
ARQ	Automatic Repeat Request.
FEC	Forward Error Correction.
TCM	Trellis Code Modulation.
PDU	Protocol Data Units.
OSIRM	Open System Interconnection Reference Module.
SDLC	Synchronous Data Link Control.
HDLC	High Level Data Link Control.
LAP-B	Link Access Procedure B.
PUC	Public Utility Commission.
SCC	Specialized Common Carrier.
VAN	Value Added Network.
BOC	Bell Operating Company.
ACD	Automatic Call Distributor.
ANI	Automatic Data Identification.
CPE	Customer Premises Equipment.
S/N	Signal to Noise Ratio.

CD	Compact Disk.
CCITT	Consultive Committee on International Telephone and Telegraph.
ADPCM	Adaptive Pulse Code Modulation.
CVSD	Continuously Variable Slope Delta.
FDM	Frequency Division Multiplexing.
SSB	Signal Side Band.
LEC	Local Exchange Carrier.
CSA	Carrier Serving Area.
ESS	Electronic Switching System.
CRT	Cathode Ray Tubes.
DTE	Data Terminal Equipment.
DCE	Data Circuit-Terminating Equipment.
EBCDIC	Extended Binary Coded Decimal Interchange Code.
LATA	Local Access and Transport Area.

ABSTRACT

Telecommunications is one of the fastest growing business sectors of modern information technologies. Today telecommunications include a vast variety of modern technologies and services. Some services, such as the fixed telephone service in developed countries are becoming mature; and some are exploding like cellular mobile communications. The new environment provides new options for users and we should be more aware of telecommunications as a whole, these services and options are explained in the project, besides that special attention I paid to the security aspects and cost of telecommunications network.

We will discuss the basic purpose of telecommunications network and see that telecommunications network consists of many different networks providing services for the users. Also the three technologies needed for communication through a network are discussed briefly, besides the three categories that networks consist of.

In general this project aims to give a basic understanding of the structure and operation of a telecommunications network. A deeper theory of telecommunications, such as the spectral analysis or signals or detailed knowledge of the operation and functions of data and voice networks.

INTRODUCTION

Telecommunications network today has become one of the most important technologies in the human beings life, since it reduces distances, cost, time and has the ability to provide information from any region connected to this network, it is also established fact that telecommunications serve as a tool for development for the whole world.

Every life is dependent on telecommunications. Each of us uses telecommunications services and services that rely on telecommunications daily, such as banking; automatic teller machines, telebanking, aviation; booking of tickets, booking hotel rooms by travel agencies and a lot of services.

This thesis aims to give answers to the fundamental questions concerning telecommunications network and services, telecommunications as a business area and the general trends of technical development. Some of these questions are: what is the structure and what are the main components of a modern telecommunications network? And what are the future developments of telecommunications network? Starting from the structure and components and ending with the future developments.

The thesis consists of introduction, three chapters and conclusion.

Chapter 1 introduces telecommunications network, an overview, historical view, the future and understanding the process of telecommunications network.

Chapter 2 is studying telecommunications network in details, the components of telecommunications network, telecommunication carriers and design philosophies.

Chapter 3 discus's the future developments of telecommunications network.

Conclusion presents the significant results, contribution and future investigation.

CHAPTER 1

INTRODUCTION TO TELECOMMUNICATIONS

1.1 Introductions and Overview

We are now in what is called the "Information Age." Information has become a commodity not only to the business community, but also to all of society. In fact, the whole economic well being of a nation may well depend on the telecommunication infrastructure in place in that nation, and the reach of that infrastructure to other nations. Just as there are techniques on how best to handle physical commodities, so there are techniques on how best to manage information, and how to transmit it in a timely fashion to where it is needed.

Everyone of us hears the word telecommunications many times everyday, but less of people who knows the meaning of this word, so let us ask, what is telecommunications?

The word communication derives from the Latin word communicare: to impart, participate. The term "information" can be viewed as a primitive (axiomatic) concept, requiring no further definition. The science of "communication" is the study of all information transfer processes.

Telecommunications has been defined as a technology concerned with communicating from a distance, and we can categorize it in various ways. Figure 1.1 shows one possible division. It includes mechanical communication and electrical communication because telecommunications has developed from mechanical to electrical using increasingly more sophisticated electrical systems. This is why many authorities such as national Post, Telegraph, and Telephone (PTTs) are involved in telecommunications by both means.

Our main concern here is electrical and bi-directional communication, as shown in the upper part of Figure 1.1. The share of the mechanical telecommunications such as conventional mail and press is expected to decrease; while the electrical, especially bi-directional, communication will increase and take the major share of overall turnover of telecommunications in the future. Hence, major press corporations are interested in electrical telecommunications as a business opportunity. Telecommunication is expected to be one of the most rapidly growing business sectors during the next few years.

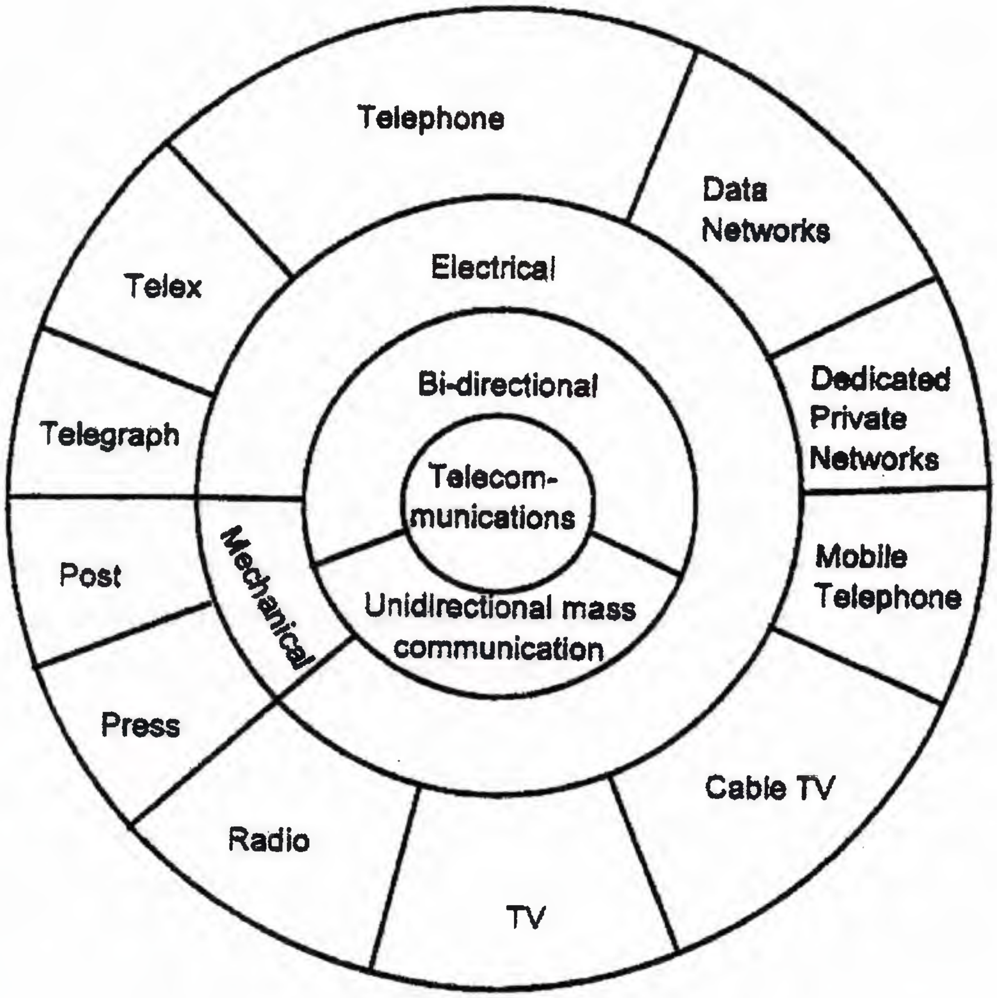


Figure 1.1 Telecommunications.

Given two entities A and B, communication is the action of transferring information from A to B, and vice versa. A and B are communicating when A is suitably able to code a message of information and relay it to B through an appropriate medium, while B is suitably equipped to receive the message by using or interpreting it in some fashion. A set of entities A, B, C, . . ., is in mutual communication when the pairs (A,B), (A,C), (B.C), etc. can ,at appropriate times, communicate with other. Protocols are agreed conventions between communicating entities on how to carry out mechanics of the communication process.

Telecommunication entails disciplines, means, and methodologies to communicate over distances, in effect, to transmit voice, video, facsimile, and computer data. Data communication entails disciplines, means, and methodologies particular to transmission of computer data, possibly over a specially engineered network, and typically from a protocol perspective. The data communication field is a subset of the telecommunication field.

A telecommunications network is the combination of numerous network elements that are required to support voice, data, or video services in local or long-distance applications. A telecommunications network is the foundation of all telephony activity; it is the network that connects the end user to virtually anywhere in the world through the use of copper cable, coaxial cable, and fiber cable or through wireless technology such as microwave or satellite. Telecommunication provides communications over a distance using technology to overcome that distance. It usually means the transmission of words, sounds, pictures, or data in the form of electronic signals or impulses, sent either as an individual message between two parties or as a broadcast to be received at many locations. While broadcasting is far removed from private communications, a new range of one-to-one communication services (including video-on-demand, and other personal information and entertainment services provided over cable networks and so-called "web casting" over the Internet) will blur the current clear distinction between the two.

Telecommunications network is a system and method for controlling on a worldwide basis two or more telecommunications networks, which are they selves capable of exercising a form of common channel signaling network control. The system uses an architecture in which a destination telecommunications network having common channel signaling control is connected to an originating telecommunications network having common signaling control through a call set up and control methodology which provides ad hoc connection between the two spaced telecommunication networks and common channel signaling networks via an unrelated world wide data network which preferably constitutes the internet.

Since its invention by Alexander Graham Bell in 1876, the telephone has become the most familiar form of telecommunications. More recently, voice telephony has been supplemented by a range of computer-based telecommunication services. These have become popular through the Internet and World Wide Web-vast computer networks that provide many people with the means to exchange information.

Telecommunication system involves Public Switching Telephone Network (PSTN), Public Land Mobile Telephone Network (PLMN), radio and television network and the emergence worldwide computer. Technological advancements develop an exponential fashion in the telecommunications industry. Interconnection Local Area Network (LAN), Metropolitan Area Network (MAN) and Wide Area Network (WAN) via PSTN has created an advanced generation of switching technologies such as Asynchronous Transfer Mode (ATM), Integral Service Digital Network (ISDN), Synchronous Optical Network (SONET), Switched Multi-megabyte Data Service (SMDS), Fiber Distributed Data Network (FDDI).

The theme "Telecommunications and the Environment" is a particularly important subject, which could hardly be more relevant to the world today.

An outsider, and even someone working in an area specifically concerned with the environment, might well ask what telecommunications and the environment have to do with each other.

At first sight, the environment, in the broadest sense of the word, would seem to be unrelated to telecommunications. And yet, there are very real links between the two. They are in fact more or less inseparable.

It is a generally established fact that telecommunications serve as a tool for development. What kind of development do we mean? The answer is clear: sustainable development. If we first consider the problem of the environment and then define the concept of sustainable development, it should be easy to demonstrate the almost organic links between telecommunications and the environment, taking a look at existing technologies, the information they convey in all its forms and the extent to which they can provide solutions to the problem of environmental protection.

The deployment of the telecommunications network is the final stage of the process, and it requires experts from a number of different disciplines, including design (outside plant and central office [CO], construction (outside-plant cable placing), and CO equipment installation and testing. The most effective way to manage this stage of the process is to use a company that serves as a single point of contact with project-management expertise and that can manage every aspect of the entire job (see Figure 1.2).

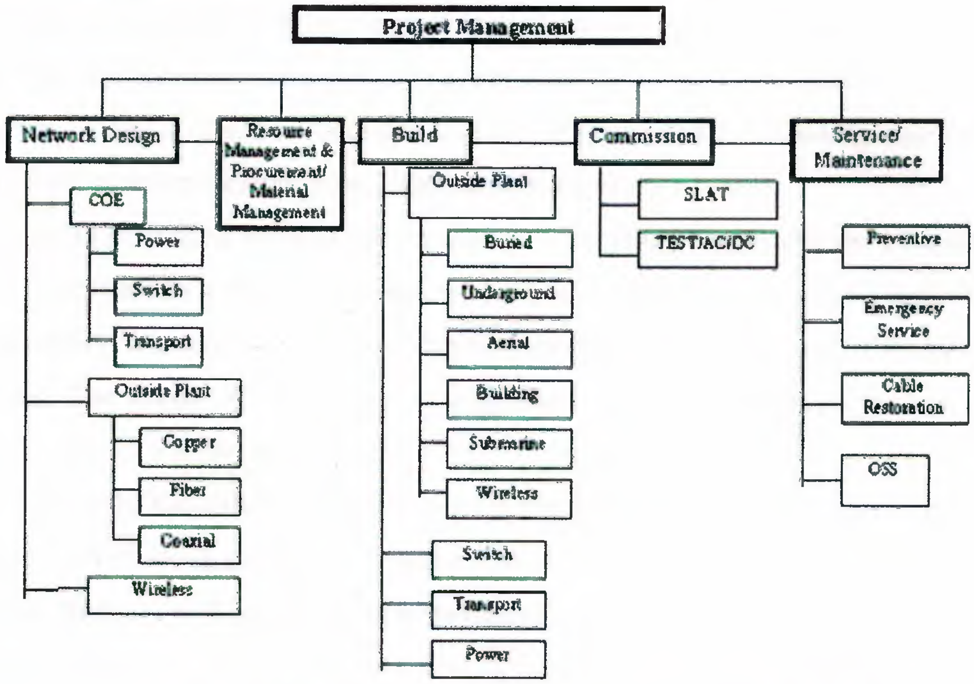


Figure 1.2 Deployment of a Telecommunications Network.

1.2 Historical View

It is now taken for granted in developed nations that by pressing a few buttons people can talk to family, friends, or business associates across the world. The technology that has led to one of the most complex creations of the 20th century, the telephone network has evolved over the past hundred years or so.

The first electrical means of communication was not the telephone, however, but the telegraph, which allowed messages, sent in code (usually Morse Code) to be received and printed at a distant location. The age of commercial telegraphy dawned in 1839 when the British pioneers William Fothergill Cooke and Charles Wheatstone opened their line alongside the main railway route running west from London. A technically simpler system of telegraphy was devised in 1843 by Samuel Morse, and after this the spread of telegraph networks was rapid, with routes spreading across most of the countries of the Old and New Worlds and then beneath the oceans that separated them. By 1930 nearly

650,000 km (400,000 mi) of undersea cables had been laid, linking the economic, political, military, and cultural institutions of the world.

An even greater breakthrough was made in 1876, when Alexander Graham Bell made the first telephone call to his assistant with the words "Mr. Watson, come here, I want you". Bell's invention sparked a series of innovations, ultimately culminating in today's information superhighway. Key steps along the way were:

In 1889 Almon Strowger developed an automatic switching system that could set up a telephone call without intervention by a human operator. Strowger's motivation for this invention was to prevent his calls being diverted to a business competitor by his local operator. The impact of the invention was much wider as it provided the basis for the current telephone network.

In 1901 Guglielmo Marconi demonstrated that radio waves could be used to transmit information over long distances when he sent a radio message across the Atlantic. Radio is still one of the key transmission media today, and is the basis of many mobile services.

In 1947 William Shockley, John Bardeen, and Walter Brattain invented the transistor. This enabled the electronics revolution to take place and provided the basis for a computerized, rather than mechanical, telecommunications network.

In 1965 Charles Kao put forward the theory that information could be carried using optical fibers. These have subsequently been developed to provide a means of carrying huge amounts of information at very high speed. Optical fibers form the backbone of the global transmission network.

1.2.1 The 1990s (The Century Closes - Into the Next Millennium)

1990- IBM sells ROLM Corporation to Germany based telecommunications giant - Siemens Corporation. AT&T develops the optical digital processor. Telmex (Mexico) is privatized. Telecom New Zealand is privatized.

1991-Bell Labs develop photonic switching. The Court of Appeals orders Judge Green to lift the ban on RBOC entry into information services. AT&T fights the long fight and buys NCR. GTE purchases Contel.

1992-The first color motion videophone introduced in the United States. The U.S. reaches its 10 millionth cellular subscriber. The Cable TV Act is introduced to regulate CATV pricing.

1993-The first digital mobile network is established in the U.S. (Los Angeles) while the first all digital cellular networks is brought up in Orlando, FL. The FCC allocates spectrum for PCS. Europe sets 1998 as the date for full liberalization of its telecom markets.

1994-The FCC begins PCS auctions. AT&T purchase McCaw Cellular. The number of subscriber telephones lines in the United States reaches 157.9 million; in the world: 609 million.

1995-There are now 25 million cellular subscribers in the U.S. Worldwide; 30 million users are now on the Internet.

1996-Commercial PCS operations begin in the U.S. The Cable Modem is introduced as the number of U.S. cellular subscribers reaches 40 million. AT&T announces its second major divestiture by spinning off NCR and its equipment business (including Bell Labs) under the Lucent Technologies name. Deutsche Telekom (Germany) is privatized. WorldCom and MFS merge to join local and long distance service providers. MCI and British Telecom merger to create Concert. France Telecom and Deutsche Telekom buy 10% of Sprint to form Global One alliance. Southwestern Bell announces merger plans with Pacific Telesis. Bell Atlantic announces plans to merge with NYNEX. The Telecommunications Act is the first real revision of the Communications Act of 1934 and is aimed at creating full competition in all markets.

1997-AT&T completes divestiture of Lucent Technologies and NCR. Implementation of the Telecommunication Act of 1996 is held up in some quarters in appeals courts as the RBOCs and the long distance companies' battle over requirements. Bell Atlantic wins approval of its takeover of NYNEX. SBC wins approval of its purchase of Pacific Telesis. The number of RBOCs now sits at five. Ameritech and BellSouth file, under the Telecom Act of 1996, to provide long distance services within their service areas. Both companies are refused permission to do so based on the lack of competition in their local markets. The RBOCs contest (and win) the FCCs authority on overseeing the opening of local telephone markets under the Telecommunications Act of 1996. British Telecom loses its bid for MCI to WorldCom. Lucent Technologies acquires Octel Communications for $1.8 billion. SBC wins a temporary victory when a Texas court rules that the Telecom Act of 1996 is anti-competitive by requiring the RBOCs to complete a series of steps to open their local markets while placing no such requirements on competitors (IXCs) wishing to enter the local markets.

1998-The FCC condemns the Texas Court's ruling and requests a decision from the Federal Courts. AT&T announces plans to acquire Teleport Communications. SBC announces its plan to acquire Southern New England Telephone (SNET) of Connecticut - in the heart of Bell Atlantic territory. AT&T announces plans to merge with TCI. WorldCom sells its Internet unit to Cable & Wireless. The WorldCom purchase of MCI is approved. SBC announces its plan to "merge" with Ameritech. Bell Atlantic announces plans to merge with GTE. Ameritech attempts to enter the long distance market using Qwest. Bell Atlantic and Bell South's bids to enter the long distance market are denied. The FCC's attempt to lower access fees and implement"universal service charges" results in higher costs to the end user and a stampede of user complaints.

1999-Organizations all over the world spend billions of dollars as they try to make their telecommunications systems and networks ready for the turn of the century. MCI WorldCom becomes official. Mergia Mania strikes the telecommunications industry as thousands of small start-up companies are purchased by larger ones. SBC and Ameritech announce plans to merge. The Internet envelopes the business community as companies scramble to ensure that they are ready to do business via this "World Wide Web". Bell Atlantic becomes the first "Baby Bell" to be approved to offer inter-LATA long distance services to customers in New York. The antiquated manner of telephone number distribution and the soaring number of competitive local exchange carriers (CLECs) places demands for new area codes throughout the country. The FCC allows the "universal service" percentage rate charged to IXC (and thus customers) to rise. Qwest and Global Crossing plans to merge fall through as Qwest moves to purchase "Baby Bell" US West. MCI WorldCom announces plans to merge with Sprint. The Federal Government continues its anti-trust suit against Microsoft Corporation.

1.2.2 The 2000s (The 21st Century Begins)

2000- Years of preparation and billions of dollars result in the Y2K "Bug" being nothing more than a minor pest on January 1st. Mergers run rampart in the telecommunications industry as companies plan for the future and the Internet. Lucent Technologies announces it will "spin off" its enterprise solution group into a new company. Bell Atlantic and GTE announce that their new combined company will be named "Verizon" (derived from the Latin for truth) thus moving the "Bell" name into its history. Cisco and Nortel networks jockey for position as leaders in voice and data communications and the Internet. Qwest divests itself of long distance companies in the US West service area in order to complete its purchase of US West. President Clinton requests that the "universal service" fee be increased to allow accommodation of Native American reservations and their technology needs. AT&T moves toward completion of its acquisition of Media One Communications. DSL becomes the new "hot service" for homes and small businesses accessing the Internet. The Federal Government rules against Microsoft Corporation - calls ring out for the organization to be divested into two separate companies. The Federal Trade Commission recommends rejection of the MCI WorldCom and Sprint merger putting this mega-deal at jeopardy. The Supreme Court upholds the FCC "detariffing" section of the Telecom Act of 1996 - All non-dominant carriers must conduct this detariffing over the next seven months (May - December). Lucent Technologies divest itself of its enterprise systems division creating Avaya Communications. WorldCom announces itself intent to buy Intermedia. British Telecom and AT&T officially call off rumored merger talks as AT&T hits financial hard times. First AT&T and then MCI announce major "spin offs" of primarily their consumer long distance divisions. Verizon wins the recommendation of Massachusetts's regulators to offer long distance service in Massachusetts; approval by the FCC is expected in December. The year turns into a financial disaster for many so called "dot com" companies as venture capital and high stock prices dry up resulting in layoffs, bankruptcies and the nickname "dot bombs". Verizon withdraws its application to provide long distance service to Massachusetts residents and companies. DSL providers nationwide fall on hard times as provisioning and maintaining the service becomes too expensive resulting in companies like Harvard Net and Digital Broadband to discontinue DSL service. SBC pays a record

$6.1 million fine for not meeting performance standards set as a condition of its merger with Ameritech.

The modern telephone network can be viewed as a globally distributed machine that operates as a single resource. Much of it uses interconnected computers. The network that most people use to carry voice traffic can also be used to transfer data in the form of pictures, text, and video images.

1.3 The Future

No one can predict with certainty exactly how telecommunications will develop, but certain trends can be noted. The cost of communication is falling in real terms, making advanced applications more affordable. Broader competition in the marketplace will reduce prices further. Telephone companies (telcos) recognize their revenues from carrying calls will decline and are encouraging, successfully, many new valued-added services that combine communication with the supply of information or services. Most of these have yet to evolve, but electronic commerce, mobile commerce, and various information-on-demand services are already being developed.

Some communication services currently provided by wire are migrating to radio means for greater convenience and flexibility; this includes not just cordless telephones in the home and the workplace but also connecting these telephones to the network.

Conversely radio and television programs, traditionally broadcast over the airwaves, are moving on to cable networks.

Fixed-mobile convergence is another trend, in which the distinction between conventional telephones and mobile networks will dwindle. Many people will carry a single "personal communicator" that functions as a cordless phone within the home, as their business extension in the workplace, and as a pocket mobile telephone elsewhere.

Today, thousands of information channels can be carried by a single underlying medium, like the fiber optic cables, which have a high bandwidth and a high transmission rate. A fiber optic link can carry multiple gigabit data rates. In the 10^{10} b/s range. An empirical review of the bandwidth that can be carried over transmission systems, shows that the bandwidth has increased by an order of magnitude every 20 years: 1950s: 10^{8} b/s; 1970s: 10^{9} b/s; 1990s: 10^{10} b/s. transmission bandwidth may increase even more in the future, particularly considering the potential of fiber optic technology. If the above empirical rule were to hold the data rate of the next few decades will be: year 2010: 10^{11} b/s and year 2020: 10^{12} b/s.

There is a significant trend toward outsourcing in the telecommunications industry. Suppliers in North America currently deal with numerous contractors and are finding that they are losing control of the costs and the quality of work. More and more, carriers are realizing the benefits of outsourcing to third parties so as to offer services such as customer-care and billing systems, network planning, and construction and operations support systems (OSSs).

Currently, 60 percent of service providers are outsourcing to third parties, but that number is projected to increase to 74 percent within two years. At present, 28 percent of service providers report that they outsource network planning and construction (i.e., deployment of the network) to third parties, and this number is projected to increase to 38 percent in the near future.

Data networks are altering the makeup of today's networks; as a result, suppliers in North America are expanding their networks to provide greater broadband to their customers. This is usually accomplished in one of two ways: through new construction or by retrofitting existing networks. Other trends include placing fiber and remotes closer to the home and upgrading switches. As a result, there are not enough installers for these areas, and existing installers cannot handle the peak load.

As the workload increases, the demand for quality installation and construction services increases as well. Using a large company with a permanent/long-term employee base as a single source supplier will ensure that the skilled manpower will be available to complete the project to the highest quality standards while controlling costs and schedule.

1.4 Understanding the Process

To appreciate the value of using a single source provider with project-management capabilities, one must understand all of the components involved in deploying a telecommunications network.

A simple analogy to describe the process for deploying an integrated telecommunications network today is that of a new subdivision that has just been developed. Within this subdivision, 500 new homes must be connected to the existing telephone network in the community. This example assumes that a remote-access node will be built within the subdivision. The process of connecting this subdivision to the local-access network switch, which connects to the long-distance access switch, which, in turn, connects to the rest of the world, is a multiple-step process (see Figure 1.3).

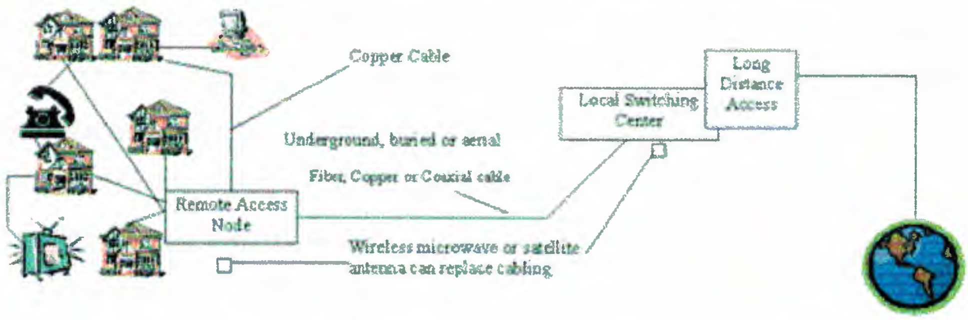


Figure 1.3 Example of Building an Integrated Telecommunications Network for a Subdivision.

Step 1: Outside-Plant Network Design (Engineering)

In keeping with the analogy mentioned earlier in the tutorial, the outside-plant designer for the project is responsible for route selection between the houses in the new subdivision and any reinforcement to the local switching center. In the design, the outside-plant designer can plan for the placement of underground, buried, aerial, submarine, or building cable installation or wireless installation, depending on a number of factors including the terrain, existing infrastructure, environment, etc. Specifically, the network designer is responsible for the following:

- Route planning
- Identifying right-of-way requirements and potential design conflicts
- Negotiating right-of-ways
- Determining specialized design, plan, and digital mapping requirements
- Preparing preliminary designs based on clients' specifications
- Developing firm price quotations based on preliminary design and estimating tables
- Providing as-built plans and specifications
- Completing final design and specifications for installation and ongoing design changes as required during installation

- Identifying material requirements and providing material ordering input to client or external suppliers as required

Step 2: CO Design (Engineering)

The design of the CO involves understanding what equipment must be installed to make the network work. Keeping with the subdivision analogy, we are assuming that there will be a remote-access node in the subdivision. In this particular example, the designer will be responsible for the following:

- Determining what equipment must be added to the existing switching center
- Determining the sizing of the access node
- Designing the transport system between the remote and the host
- Determining the digital equipment and transport system that will be used between the switch and the remote
- Reviewing the power system to see if it must be reinforced (power study)

Step 3: Outside-Plant Construction

Once the design and specifications have been determined, the installation of the cable (copper, fiber, or coaxial) must be installed to those specifications. This work is generally performed by highly skilled splicing and line technicians who are qualified to place and connect cable in a variety of outside-plant networks, including live circuits. Their responsibilities also include testing continuity and troubleshooting in existing networks. If it is determined that wireless technology is to be used, the infrastructure (i.e., towers) are built at this stage.

Step 4: CO Equipment Installation

The next step is to install and commission the specialized equipment to make the whole network work. Most CO and switching equipment is housed in a localized switching center or access nodes (remote switch) that is located within the subdivision that links back to the switching center. The CO equipment (COE) technicians are suppliers trained to install the specialized equipment that routes the calls to the appropriate switch. Some of the many types of equipment that must be installed and maintained by these technicians are as follows:

- Switching equipment including Nortel digital multiplex system (DMS) technology
- Transport equipment such as channel banks, fiber multiplex transport (FMT), digital access and cross-connect system (DACS), New bridge, and various miscellaneous peripherals (i.e., asynchronous transfer mode [ATM], frame relay, and network-management hardware)
- Access remote includes remote carrier urban (RCU) 600/900, DMS–1U, remote switch concentrator (RSC), and remote line concentrating module (RLCM), which are installed into various walk-in cabinets and environmentally controlled manhole enclosures
- FMS equipment patch panels, routers, bridges, and active hubs.
- Powers The technicians regularly install, replace, and upgrade rectifiers, inverters, batteries, and mechanized frame administration (MFA) power plants. They also install grounding into COs, access nodes, and customer-owned telephone rooms that are required to meet the grounding standards.
- Synchronous optical network (SONET) transport—The COE technicians are also experienced in building large Internet protocol (IP) networks and are supplier-trained to do system lineup and test (SLAT), including software upgrades on live equipment and optimization. They are also trained in optical carrier–3 (OC–3), OC–3E, OC–12, OC–48, OC–192, access nodes, and access node express.

Step 5: Commissioning

The commissioning of the newly installed network involves testing to make sure the network is up to specifications before it is turned on. Once it is determined that everything is operating according to specifications, it will be integrated into the live network.

Chapter two introduces the main components of the telecommunications network, which are the voice communication and data communication, and it studies the transfer of signals, switching and signaling.

CHAPTER 2

TELECOMMUNICATIONS NETWORK

2.1 An Overview

This chapter describes the basic operation of a telecommunication network with the help of a conventional telephone. The operation of a conventional telephone, which is easy to understand, is used to clarify how telephone connections are built up in the network. We look at the subscribers signaling over the subscriber loop of the telephone network. The same main signaling phases are needed in modern data and mobile networks. We start with this simple service to lay a foundation for understanding more complicated services. In this chapter we divide the network into layers and briefly describe different network technologies that are needed to provide various kinds of services. Some of these, such as mobile and data networks and their services. One of the topics which is introduced in this project is the theory of traffic engineering, that is how much capacity we should build into the network in order to provide a sufficient grade of service for the customers.

2.2 Basic Telecommunications Network

The basic purpose of telecommunications network is to transmit user information in any form to another user of the network. These users of public networks, such as a telephone network, are called subscribers. User information many have many forms, such as voice or data, and subscribers may use different access network technologies to access the network, such as fixed or cellular telephones. We shall see that telecommunications network consists of many different networks providing different services, for example, data, fixed, or cellular telephony service. These different services are discussed briefly due to the next three sections and approximately full explanation due to the chapter. In the following sections we introduce the basic functions that are needed in any networks regardless of what services they provide.

The three technologies needed for communication through a network are:

- Transmission;
- Switching;
- Signaling.

Each of these technologies requires specialists for engineering, operation, and maintenance.

2.2.1 Transmission

Transmission is the process of transporting information between end points of a system or a network. Transmission systems use three basic media for information transfer from one point to another:

- Copper cables, such as LANs and telephone subscriber lines;
- Optical fiber cables, such as high data-rate transmission in a telecommunications network;
- Radio wave, cellular phones and satellite transmission.

In telecommunications network the transmission systems interconnect exchanges, and these transmission systems altogether are called the Transmission or Transport Network. Note that the number of speech channels (which is one measure of transmission capacity) needed between exchanges is much smaller than the number of subscribers since only a small fraction of them has a call connected at the same time.

2.2.2 Switching

In principle all telephones could still be connected by cables as they were in the very beginning of the history of telephony. However, when the number of telephones grew, it was soon noticed that it was necessary to switch signals from one wire to another. Then only a few cable connections were needed between exchanges because the number of ongoing calls was much smaller than the number of telephones; see Figure 2.1. The first switches were not automatic and switching was done manually on switchboards.

Automatic switches, known as exchanges, were developed in 1887 by Strowger. Then the witching had to be controlled by the telephone user with the help of pulses generated by a dial. For many decades exchanges were a complex series of electromechanical selectors, but during the last twenty years thy have developed into a software-controlled digital exchanges that can provide additional services. Modern exchanges usually have quite a large capacity, tens of thousands subscribers, and thousands of them may have an ongoing call at the same time.

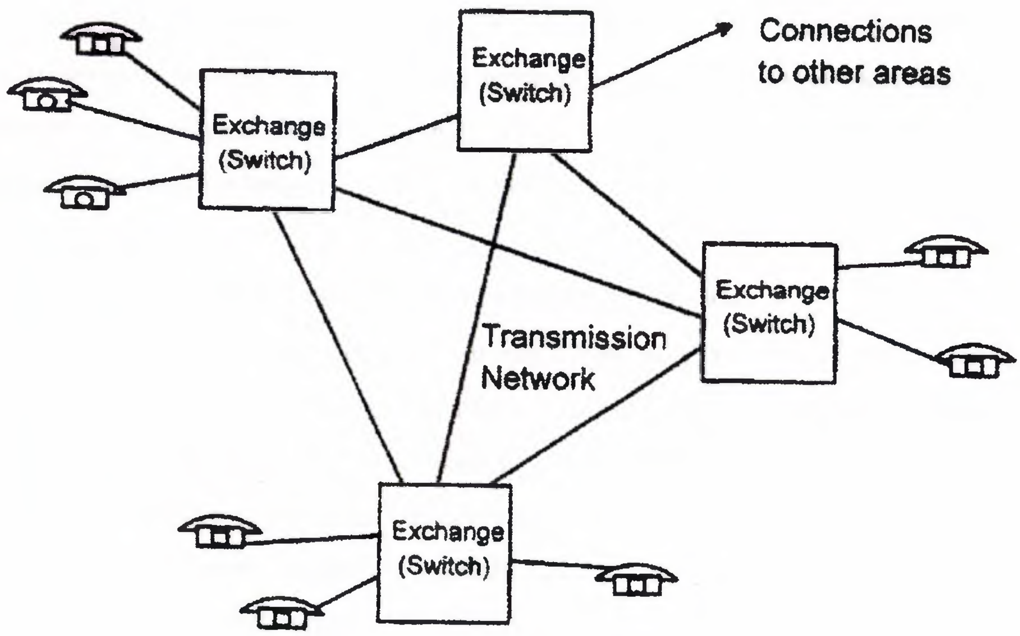


Figure 2.1 A Basic Telecommunications Network.

2.2.3 Signaling

Signaling is the mechanism that allows network entities (customer premises or network switches) to establish, maintain, and terminate sessions in a network. Signaling is carried out with the help of specific signals or messages that indicate to the other end what is requested of it by this connection. Some examples about signaling examples on subscriber lines are:

- Off-hook condition: the exchange notices that the subscriber has raised the telephone hook (DC-loop is broken) and gives a dial tone to the subscribers.

- Dial: the subscriber dials digits and that are received by the exchanges.
- On-hook condition: the exchange notices that the subscriber has finished the call (subscriber loop is connected), clears the connection, and stops billing.

Signaling is naturally between exchanges as well because most calls have to be connected via more than one exchange. Many different signaling systems are in use for the interconnection of different exchanges. Signaling is an extremely complex matter in a telecommunications network. Imagine, for example, a foreign GSM subscriber switching his telephone on in Hong Kong. In a few seconds he is able to receive calls directed to him. Information transferred for this function is carried in hundreds of signals of signaling messages between exchanges in international and national networks.

2.3 Components of a Telecommunications Network

In this section, we examine various types of networks. A telecommunication network can be viewed as an ensemble of a number of links, such as those shown in Figure 2.2. Telecommunication networks consist of three general categories of equipment: termination equipment, transmission equipment and switching equipment. Each of these three categories, in turn, comprises a number of subcategories or technologies. Figure 2.3 depicts a typical telecommunication network of the 1990s. As we can see, these networks can be very complex and many employ a variety of technologies.

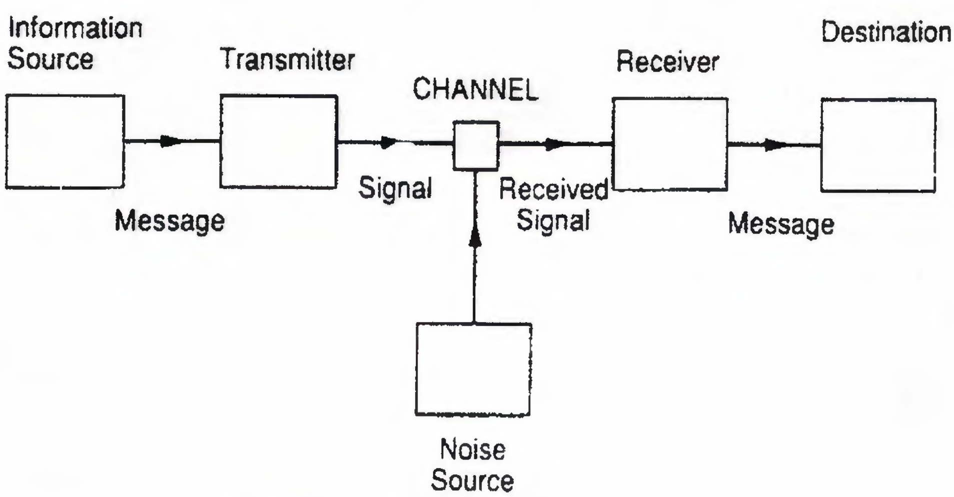


Figure 2.2 Communication Model.

Network can be public or private. Public networks can be used by everyone, similar to public transportation. Private networks are typically restricted to one organization (or a group of organizations); the general public is not allowed access to a private network, just as one can not employ (without permission or by arrangement) somebody else's private automobile. Both types of networks basically have the same functionality and capabilities, although different capabilities may be accentuated in one or the other. Public networks are provided by common carriers.

Public networks usually utilize carrier-provided switches, also known as exchange. Prior to the bell system divestiture, the distinction between exchange access and interexchange communication was not a matter of major regulatory importance, except possibly in terms of the rates and tariffs. After divestiture, terms such as "local exchange" and "interexchange" acquired an additional legal distinction; local exchange access service and interexchange service (more specifically, LATA-local access and transport area) must be provided by separate entities, and the old Bell companies are currently precluded from offering domestic interexchange services (with minor exceptions in some major corridors, for example, Northern New Jersey and New York City).

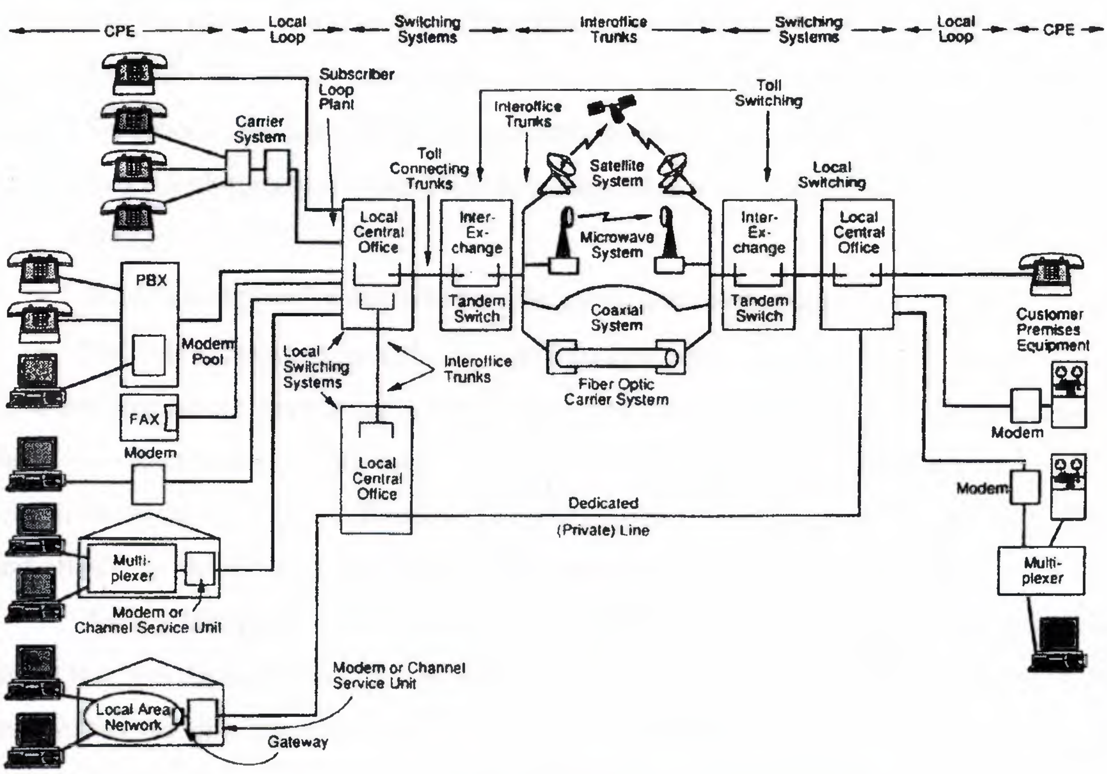


Figure 2.3 A modern telecommunication network.

2.3.1 Voice Networks

The traditional public switched telephone network was originally developed to service voice traffic. Data can also be carried by the same network when a modem (modulator-demodulator) is employed by users at each end of the link. In effect, the modem transforms the data into an acoustical signal that fits into the nominal 4-kHz bandwidth of a standard telephone channel. This method of carrying data is called voice band or circuit-mode data. Improved network facilities more suited to carrying data in their native digital mode are now beginning to emerge.

2.3.1.1 Customer-Premises Equipment

Customer-Premises Equipment (CPE) is equipment that is owned and maintained by the user. It includes the signal-entry termination equipment (called station in the voice environment), concentration equipment, in-building or in-campus wiring, or even an entire sub network. In the latter case, the user may own the communication network up to demarcation point (the point between the public network and the user's network), or in some cases the user can also own the long-haul transmission and switching equipment.

Typical CPE in the voice area includes:

- Telephone sets (particularly after computer Inquiry II, 1980, and divestiture, 1984);
- Key telephone equipment, including telephone sets, wiring, and other components;
- PBXs;
- Inside wiring (including wire closets, jacks, and connectors);
- Recording, answering, and voice mail equipment.

CPE termination equipment (as compared to other types of CPE) accepts the user's voice, data, or video signals and encodes them so that they can be transmitted over a telecommunication network. Examples of termination CPE include voice station set, CRTs and other data terminal, facsimile machines, and video conferencing cameras.

Analog termination devices encode the user signal (i.e., speech) in to an electrical signal that replicates the energy content of the original signal. This electrical signal is then transmitted to the desired remote location. Intuitively, digital termination devices "measure" the height of the signal at frequent intervals, and then represent that height with a binary coded number (more sophisticated methods perform signal analysis rather than simple energy measurement). For a computer terminal, the signal is already in digital form. The digital representation of the signal is then transmitted to the desired remote location, where it is utilized by the receiving CPE in digital form, or is converted into analog form. As an alternative, the user's CPE delivers an analog signal to the network; the network transforms this signal into a digital stream for more reliable transmission, and then reconverts it into an analog signal for delivery to the intended destination.

Binary numbers, employed in digital transmission, are composed of combinations of 0s and 1. The individual 0s and 1s are called bits. A collection eight bits is called octet (less precise term is byte). Sometimes octets and bytes are also called words, although the term is more appropriate to describe a collection of octet. For example a 32-nit computer is said to have 32-bit words (four octets). Digital communication channels are measured in terms of their information carrying capacity in b/s.

Initially, the public telephone plant was an analog network, optimized for analog voice transmission. This included analog transmission and analog switching facilities. Beginning in the early 1960s many interoffice trunks (links between switches) began to be replaced with digital links. In the mid-1970s, switches also began to handle digitized voice directly (i.e., without multiple conversions between analog and digital). Today digital links are common and prevalent. New network architecture, ISDN, aims at providing end –to-end digital circuits to the customer. All major telecommunication carriers in the United States and abroad have stated that ISDN is the strategic direction of their networks. Digital circuits are more suited to data transmission applications than are analog circuits.

(a) Voice Digitization Schemes

The telephone instrument performs the function of coding the user's voice into a signal suitable for subsequent transmission over the network. For public switched networks, an evolution is taking place in this area, tracking (sometimes leading) a similar evaluation in private networks. The evaluation is as follows. Until 1960s, the telephone set generated an analog signal, which was transmitted through the network in an analog fashion, end-to-end. Beginning in the 1960s and continuing through the 1980s, while the set still generated analog signal, the voice could be digitized in the transmission or in the switching components of the network {first in time in trunks between central offices (COs), 1962; then in the loops between the user and the CO, 1973; then in the proximity of the switch itself, so that a digital switch could terminate under multiplexed loop or trunk carrier systems, 1976}. With ISDN in the 1990s, the telephone set will be allowed to generate a digital signal representing the user's voice, for end-to-end transmission in digital form. (In a number of PBXs, this digitization at the telephone set is already taking place).

Historically, digitization techniques have been identified with activities performed in the network, particularly in reference to trunk carrier systems, digital loop carriers, and digital switches, as described in the previous paragraph. Because the future belongs to ISDN (or, at least, to digital communication), voice digitization techniques are here properly seen from a CPE perspective (the continued introduction of CPE high speed multiplexers also support this perspective). The digitization techniques do not change this change in perspective; the perspective only determines in what context the subject is treated. Remember, however, in spite of the present perspective, that as of 1990, a very large percentage of the voice digitization still occurs within the network proper.

To digitize the voice means to represent it with a stream of numbers coded in binary representation. Two classes of methods are used to digitize voice: waveform coding and vocoding. In waveform coding, we attempt to code and then reproduce the analog voice curve by modeling its physical shape. The number of b/s to represent the voice with this method is high: 64, 32, 16, or at least 9.6 kb/s, depending on the technology. Vocoding attempts to reproduce the analog voice curve by performing a mathematical analysis (fast Fourier transform) that "identifies" abstractly the type of curve; what is transmitted is a small set of parameters describing the nature of the curve. The number of kb/s to represent the voice with this method is low: 9.6, 4.8, 2.4, and even 1200 b/s, depending on the technology. Voice quality is increasingly degraded as the digitization rate becomes smaller. An extensive body of research on vocoding methods has evolved in the past 15 years (at least count more than 700 technical papers have been written).

Digital speech quality through a network is also degraded by the accumulation of quantization noise introduced at signal conversion points; conversion can occur several times in a network, in partially digital environment. In a nearly totally digital environment, only one analog-to-digital conversion close to the source and one digital-to-analog conversion close to the destination are required. In a totally digital environment, the conversion takes place right at the source, and not in the network. Voice quality will improve substantially in these environments, particularly with the high-quality coding.

(b) Pulse Code Modulation

The simplest waveform method to convert analog speech to a digital stream is process called pulse code modulation (PCM). PCM was invented in the 1930s, but only become prevalent in the 1960s when transistors and integrated circuits become available.

Nyquist theory specifies that to code properly an analog signal of bandwidth W with basic PCM techniques, we need 2W samples per second. For voice, band limited to a nominal 4000-Hz bandwidth, we need 8000 samples per second (the actual telephony frequency range used in 300 to 3400 Hz). The dynamic range of the signal {and ultimately the signal to noise ratio (S/N)} dictates the number of quantizing levels required. For telephonic voice, 256 levels suffice, based on psychoacoustic studies conducted in the 1950s and early 1960s, it follows that 8 bits are needed to represent many levels uniquely. This, in tern, implies that we need 64,000 b/s to encode telephonic human speech in digital form. PCM does not require sophisticated signal processing techniques and related circuitry; hence, it was the first method to be employed, and is the prevalent method used to day in telephone planet. PCM provides excellent quality. This is the method used in modern compact disc (CD) music recording technology (although the sampling rate is higher and the coding words are longer, to guarantee a frequency response to 22kHz). The problem with PCM is that it requires a fairly high bandwidth (64 kb/s0 to represent a voice signal. PCM is specified by the CCITT's (Consultative Committee on International Telephone and Telegraph) Recommendation G.711. The CCITT is a standards making body. Two "laws" (recommended standards) describe voice compression in PCM: in the United States, the μ-law is used; in Europe, the A-law is employed. The reason to follow specific PCM standards is that we want to be able to install equipment from different manufacturers and still retain system integrity and compatibility. (Although PCM can be mathematically treated as a type of modulation-which we will discuss later-many people today view it as an example of signal processing; this is the perspective we will use herewith).

One key issues is the spacing in the signal amplitude postulated by the sampling codec (also known as quantizer) to establish the boundaries where the different levels are declared. If we divide the maximum amplitude in 256 equal intervals, voice, which normally has numerous low-level signal components, would not be coded adequately. Instead, the amplitude space is sub divided with logarithmic spacing with respect to the signal origin, this affords a stable S/N ratio over a wide range of voice levels. Note, that if the input signal amplitude exceeds the maximum quantizer level, the result is clipping distortion. A quantizer must be designed to avoid frequent clipping; hence. The quantizer's maximum level is determined by the power of the strongest signal that the quantizer must handle. A signal-to-distortion ratio (S/D) of around 35 decibels is desired for a wide range of input level.

(c) Newer Coding Schemes

PCM has been around for a quarter century, and new technologies are beginning to demand attention. Sophisticated voice coding methods have become available in the past decade due to the evaluation of VLSI technology; 64 kb/s PCM is no longer the only available technique. Coding rates of 32,000 b/s, 16,000 b/s, and even "vocoder" methods requiring 4800 b/s, 2400 b/s, and even less, have evolved (intelligibility, but not speaker recognition, can still be obtained at 800 b/s).

Some interest exists in pursuing these new coding schemes because the implication is that we cam double or quadruple the voice carrying capacity of the network in place without the introduction of new transmission equipment. Of all available schemes emerging from the laboratory the adaptive pulse code modulation (ADPCM) schemes is the most promising at this time. It effectively provides "toll quality" voice with minimal degradation at 32 kb/s. the CCITT studied this algorithm and recommendation (G.721) followed in 1988. A problem with this method has been that of "passing" data at various speed under these coding methods; a number of widely deployed U.S. modems (in particular, the Bell 202 type, at 1200 b/s in half-duplex mode) fail to transmit through a digital carrier system equipped with the 32 kb/s line cards. Algorithmic refinements to deal with the problem involve fine-tuning some of the parameters that characterize the coding scheme. Performance of the coding scheme revolves around the following parameters: frequency response and tracking, idle circuit noise, transient response and warmup period, single frequency distortion, intermodulation distortion, and S/D. A standard for a 16 kb/s coding scheme and a proposal for an 8 kb/s scheme has been studied by CCITT study group XV. A brief description of the ADPCM follows.

(d) Differential PCM

If a signal has a high correlation (exceeding 0.5) between adjacent samples, as is the case for speech sampled at the Nyquist rate, the variance of the difference between adjacent samples is smaller than the variance of the original signal. If this difference is coded, rather than the original signal, significant gains in S/D performance can be achieved (conversely, a given S/D can be achieved with fewer quantizer bits). This implies that, for the same desired accuracy, fewer bits are needed to describe the change value from one sample to the next than would be needed to describe the absolute value of both samples. This is the idea behind differential PCM (DPCM). DPCM systems are based primary on a 1952 patent by Cutler.

In a typical DPCM system, the input signal is band-limited, and an estimate of the previous sample (or a prediction of the current signal value) is subtracted from the input. The difference is then sampled and coded. In the simplest case, the estimate of the previous sample is formed by taking the sum of the decoded values of all the past differences (which ideally differ from the previous sample only by quantizing error). DPCM exhibits the greatest improvement over PCM when the signal spectrum is peaked at the lower frequencies and rolls off toward the higher frequencies.

The problem with this voice coding method is that if the input analog signal varies rapidly between samples, the DPCM technique is not able to represent with sufficient accuracy the incoming signal. Just as in the PCM technique, clipping can occur when the input to the quantizer is too large; in this case, the input signal is the change in signal from the previous sample. The resulting distortion is known as slope-overload distortion.

(e) Adaptive DPCM

In adaptive DPCM, the coder can be made to adapt to slope overload by increasing the range represented by the encoded bits, which here number 4. In principle, the range implicit in the 4 bits can be increased or decreased to match different situations. This will reduce the quantizing noise for large signals, but will increase noise for normal signals; so, when the volume drops, the range covered by the 4-bit signal drops accordingly. These adaptive aspects of the algorithm give rise to its name. ADPCM transmits 4 bits per sample for 8000 samples per second, for a bandwidth of 32,000 b/s.

In practice, ADPCM coding device accepts the PCM coded signal and then applies a special algorithm to reduce the 8-bit samples to 4-bit words using only 15 quantizing levels. These 4-bit words no longer represent sample amplitudes; instead, they contain only enough information to reconstruct the amplitude at the distance end. The adaptive predictor predicts the value of the next signal on the level of previously sampled signal. A feedback loop ensures that voice variations are followed with minimal deviation. The deviation of the predicted value measured against the actual signal tends to be small and can be encoded with 4 bits. In the event that successive samples vary widely, the algorithm adapts by increasing the range represented by the 4 bits through a slight increase in the noise level over normal signals.

(f) Lower Rate Voice Digitization

Some CPE equipment (for example, T1 multiplexers) now use continuously variable slope delta (CVSD) to achieve voice digitization rates below 32,000 b/s. to understand CVSD, we consider a form of DPCM where the length of the digital word per sample is a single bit. With such a small digital word, more samples compared to PCM-DPCM can be sent in the same bandwidth. Clearly, 1-bit words can not measure loudness; hence, rather than sending the change in height of the analog signal curve, the 1-bit CVSD data refer to a change in slope (steepness) of the analog signal curve. At the sending end, CVSD compares the input analog voltage with a reference voltage: if the signal is greater than the reference, a "1" is sent and at the same time the slope of the reference is increased; if the input is less than the reference, a "0" is sent and at the same time the slope of the reference is reduced. CVSD attempts to bring the reference signal in line with the incoming analog signal. The steeper the slope (positively or negatively), the larger the output changes between samples; the CVSD algorithm increases the size of the step taken between samples each time the slope change continues in the same direction. This is similar in concept to the adaptive nature of ADPCM: a series of 1s produce a progressively larger increase in the output.

The receiver will reconstruct the sender's reference voltage, which, after being filtered, should be a replica of the original input. While one normally employs the CVSD at 32 kb/s, the actual digitization rate can be selected by the user in some systems (with ensuing voice quality implications). CVSD can operate from 64 to 9.6 kb/s. the quality deteriorates as the bandwidth decreases; this is the method employed by systems that provide 16 kb/s voice rates (the speaker still recognizable at 16 kb/s, and speech is still intelligible at 9.6 kb/s). A typical equipment line card converting four analog voice channels into four 16 kb/s digitized speech streams and multiplexing them onto a 64 kb/s channel costs around $4000 in 1990.

(g) High-Quality Telephony

Recently, the CCITT recommended an international standard (G.722/G.725) for coding wideband speech and music (50 Hz to 7 kHz at 3-dB attenuation) at 64 kb/s. this frequency range, extended at both the high end and the low end, considerably improves telephonic voice quality over the existing norm, approaching the quality of a typical car's FM radio. Extending the cutoff frequency from 300 to 50 Hz improves the naturalness of the audio signal. Some applications for these coders are in high-grade telephones in ISDN and for teleconferencing applications. In audiovisual conferencing applications, we would like to approach the quality of face-to face communication.

The codec performance requirements for voice band data are substantially different from those of voice signals. If the codec is required to encode voice band data signals as well, its cost and complexity increase. If the codec is not required to carry data, it can be optimized for best performance on speech signals. Subband coding techniques separate the signal into components occupying contiguous frequency bands, and encode the components separately. With the audio signal subdivided into two 4-kHz bands, a high S/N in the lower band becomes perceptually more important than in the higher band. An advantage of a design that uses two equally wide subbands is that each component can be subsampled to 8-kHz and the total transmission rate may be reduced in 8 kb/s steps by reducing the number of bits assigned to samples in one or the other band. A typical wideband coder accepts a 16-

kHz sampled input signal and split it into two 4-kHz (80 kHz sampled) bands using a quadrature mirror filter. The two 8-kHz sample rate subband signals are then encodec using an ADPCM coder. The upper band is encoded using 2 bits, while the lower band is allocated various bits, depending on the desired overall rate: 6 bits are used for 64 kb/s, 5 bits for 56 kb/s, and 4 bits for 48 kb/s. the two lower rates allow simultaneous transmission of an 8 kb/s (6.4 kb/s for data and 1.6 kb/s service channel) and 16 kb/s (14.4 kb/s for data and 1.6 kb/s service channel) data stream, respectively, in addition to the voice, on a signal 64 kb/s (ISDN) channel.

(h) Voice Digitization Summary

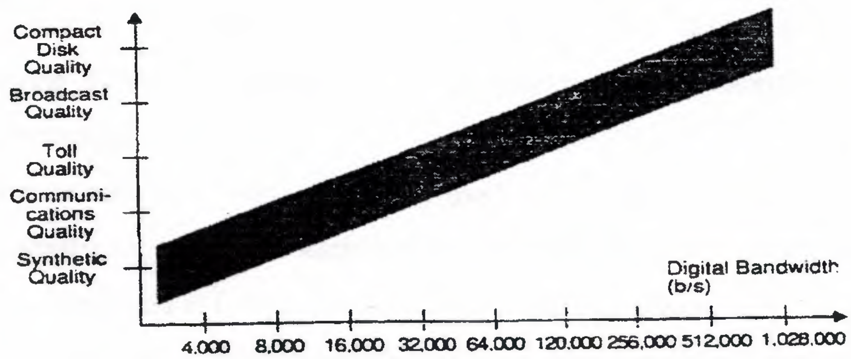


Figure 2.4 Digital voice qualities as a function of the digitization rate.

Figure 2.4 summarizes the quality versus digitization rate relationship associated with various voice coding schemes.

2.3.1.2 Transmission Equipment

The process of moving information from point to another is called transmission. The undertake transmission, one needs a variety of facilities. Transmission equipment typically includes terminal equipment (not to be confused with CPE termination equipment), which accepts the user's signal and changes it appropriately, and an interconnection medium, such as copper wire or coaxial cable among others. To superimpose the user signal onto the medium, the transmission equipment needs to modulate a carrier signal. Additionally, the equipment may multiplex a larger number of users over the same physical medium. (Multiplexers can also be CPE, if desired or appropriate).

(a) Multiplexing Schemes

A number of multiplexing schemes are available to place multiple calls in a standardized fashion on one medium. The basic multiplexing schemes are:

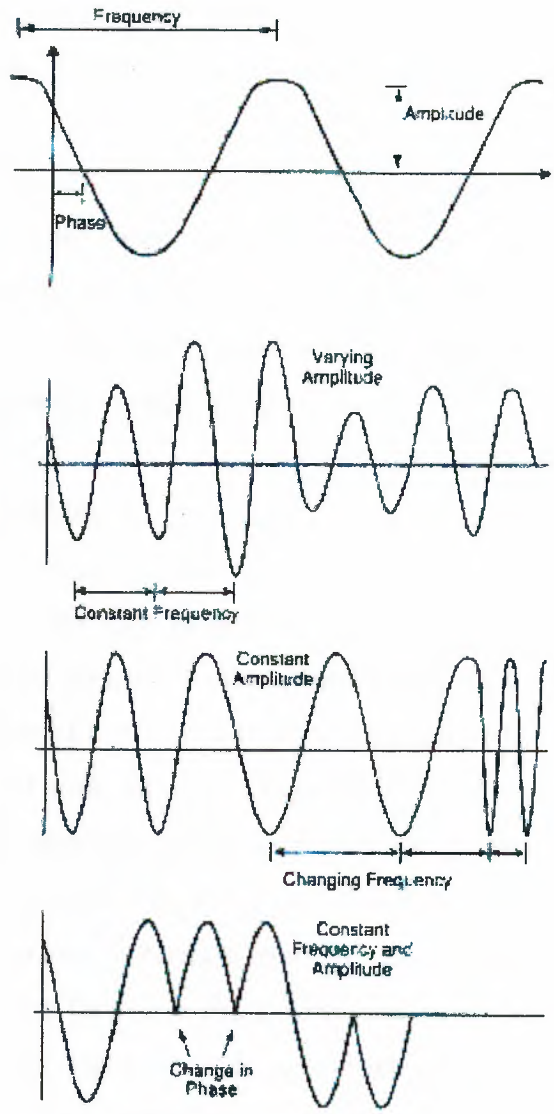
- Frequency division multiplexing (FDM). This is atypical of analog coaxial, microwave, and radio systems.
- Time division multiplexing (TDM). This is typical of digital transmission; it lends itself well to computer interfaces. In the traditional telephone network TDM has been used in conjunction with PCM coded signals.
- Space division multiplexing. An example is the frequency reuse in a cellular system or satellite.
- Code division multiplexing. Systems where the multiplexing is achieved by employing different data-stream coding methods. Used principally by military communication systems.
- Random access techniques. A method used in conjunction with TDM in which the multiplexing occurs via statistical bid and assignment of the channel. This is typically employed in LANs that do not use token disciplines.
- Demand assignment techniques. Bandwidth reservation-based systems; used in conjunction with random access. This approach is typical of satellite systems.

(b) Modulation

At the functional level, modulation is the process of imparting an intelligent signal onto an underlying carrier signal so that it can then be transmitted over a distance. The carrier signal depends on the media at hand (copper, microwave, fiber, etc.). Modulation is very common: radio and television, to mention only two obvious communication systems, employ modulation. Modulation functions come into play across the network for all types of transmission systems. The function of the modulator is to match the encoder output to the transmission channel. Figure 2.5 depicts the three characteristics of a electrical sinusoidal carrier, typical of media such as copper, coaxial, and radio: the amplitude, the frequency, and the phase. All three of these factors can be affected in order to achieve modulation. The three types of modulation are: amplitude modulation, frequency modulation, and phase modulation. The change to the carrier under each of the three methods is also shown in Figure 2.5. The modulating signal (the intelligence to carry) can be analog or digital.

Amplitude Modulation. A carrier's amplitude represents the instantaneous strength of the signal, as depicted in Figure 2.5. A carrier's amplitude of the modulated (AM), varies according to the amplitude of the modulating signal, while keeping the frequency constant. The modulation process produces a power spectrum that is symmetrical with respect to the frequency of the carrier: when viewed in the frequency domain, the modulated signal will have power spectral lines at sums and differences of the carrier frequency with the frequencies of the modulating signal. Although a major potion of the transmitter's power remains at the carrier frequency, amplitude modulation shifts energy into the sideband frequencies. This energy in the sidebands is what allows the remote end to demodulate the original intelligent signal.

Signal-sideband (SSB) modulation, an improvement of AM, concentrates most of the energy of the transmitter into the intelligence-bearing portion of the signal, enhancing the receiver signal. Because the upper and lower sidebands in AM modulation contain redundant information, one of the bands can be suppressed, after the modulation stage, giving rise to a "single sideband". This differs from AM, which uses transmitter energy to feed the carrier frequency and adds little to the intelligence received at the far end, as indicated above. SSB modulation results in a signal that requires reduced transmission bandwidth, in effect, allowing more intelligence to be transmitted over the same channel. The carrier signal can also be partially suppressed so as to use less power. This technique is often employed in microwave systems.



Top: Three characteristics of a carrier signal, Second from top: Amplitude Modulation, Third from top: Frequency Modulation, Bottom: Phase Modulation.

Figure 2.5 Modulation of the carrier

Frequency Modulation. Frequency modulation (FM) was developed in 1930s as an improvement over AM to provide high-quality music broadcasting. With FM, intelligence is added to the carrier wave by varying the frequency of the carrier in step with the frequency of the intelligence signal, while holding the output power of the carrier constant. FM is more immune to major sources of noise than AM because the most common type of noise tends to affect the amplitude. However, a frequency-modulated signal needs more bandwidth as compared to AM; even narrowband FM requires nearly twice the transmitter bandwidth of AM. FM is still extensively today for analog microwave transmission systems.

Phase Modulation. Phase modulation (PM) is similar in some respects to FM. The phase the transmitter is increased or decreased in accordance with the modulating intelligence signal. However, because small changes in phase are difficult to detect, PM is not generally used for analog applications. PM is used more commonly in digital modulation, for data transmission applications.

2.3.1.3 Subscriber Loop Plant

CPE equipment normally is connected to remote locations through a telephone company's CO. the subscriber loop is the physical link by which customers are connected with the telecommunication network. In the United States, local loops are usually provided by local exchange carriers (LECs). A star topology is employed, with loops emanating from the central point, the CO, to all local users of the network. This not only facilities management and trouble shooting of the loops but also allows central switching. Loops may be discrete two-wire copper facilities from the CO to the user, or may be partially multiplexed over a few miles of common medium. (Local loops for special services, such as data communication, may be four-wire). In this context, transmission equipment consists of the underlying physical channel, such as twisted pair, coaxial cable, optical fiber, microwave, and others, and multiplexing equipment such as digital loop carriers. To gain a perspective, we note that long-haul circuit mileage is only around 10% of the total U.S. telephone plant circuit mileage; 90% is the local loop, where the LECs install nearly 200 million miles of copper wire a year.

The physical infrastructure supporting local loops is known as outside plant. The outside plant includes conduit, poles, cable protection devices, terminals, aerial drop wire, feeder cable, and distribution cable. The cable may be strung from poles, buried, or placed in conduit. The cable is often buried underground without conduit because this approach is usually less expensive than housing the cable in conduit. The local loop is the part of the circuit that traditionally has been most susceptible transmission impairments; therefore, design and testing are both very important to maintaining quality. In designing a loop cable's characteristics, loop resistance and capacitance of the pair must all be carefully considered by the telephone company's engineers. Today's telephone networks are built around carrier serving areas (CSAs), and are served by a combination of digital or analog transmission and switching equipment. The radius of CSAs was originally defined by the bandwidth-distance performance characteristics of copper; typically, this radius is 12,000 feet. Fiber systems may eventually change the CSA concept.

The local loop infrastructure is made up of two components: the feeder plant and the distribution plant. The feeder plant provides (but not in every case) what are called "carrier facilities"-multiplexing-transmission facilities in which a number of customers may be multiplexed onto a single transmission medium. The number of multiplexed channel can vary from 24 to 96, or even to 672 or more. The LEC can combine many user channels with the multiplexing equipment, and deliver the signal to the CO where it is demultiplexed and fed to the switch for further handling. (In newer switched, it can be fed directly to the switch without demultiplexing equipment). The advantage of multiplexing is that construction of the physical plant is typically the most expensive component of the telecommunication networks (often more expensive than the electric equipment itself); thus, one of the objectives is to minimize the number of discrete physical channels needed, two methods of multiplexing in the feeder plant are frequency division multiplexing (analog) and time division multiplexing (digital). Time division techniques are becoming prevalent because of the increased deployment of digital networks and the lower cost to achieve multiplexing compared to analog techniques. The equipment at the remote end of the feeder plant also typically provides the voice digitization function for the group of stations being served.

The early examples of the feeder plant (through the 1960s) did not necessarily involve the use of multiplexing equipment. The only shared facilities in this case would be underground conduit or the poles. The distribution plant would be put in place once, to reflect anticipated growth in the sector served; the resulting junction between feeder and distribution plant occurred at a pedestal box or pole-mounted cross-connect box. In the late 1950s and 1960s, carrier systems grew in importance, due to the increased cost-effectiveness of remote electronics; an objective was set to achieve 30% of the feeder plant using carrier. At the interface, several lines can be integrated onto a single link (fiber or copper carrier) through multiplexing. The economics of scale established in the feeder plant reduce wire and cable costs that outweigh those of multiplexing. In the contemporary context, the feeder plant is largely implemented using carrier systems, and an increasing proportion of all loops are multiplexed.

The distribution plant is that portions of the network between the feeder termination (normally, some type of channel bank or remote switching module a satellite CO) and the customer. The distribution plant is generally copper-based in interoperate with the existing generation of telephone sets. Existing telephone sets require a current to operate the bell; the equipment at the feeder termination typically provides the appropriate signaling to the stations over the distribution plant. Two-wire twisted-pair cable (sometimes four-wire) is used in the distribution plant. Ideally the distance between the users and the feeder plant interface should be optimized to reduce the total length of wire required.

Fiber facilities are being increasingly introduced in the feeder plant. The only interfaces of the feeder plant are the (digital) switch at one end, and the carrier electronics at the other end. The feeder plant is thus amenable to fiber replacement without requiring additional changes to the network. Cable ducts supporting the feeder plant are overcrowded, and therefore offer ideal opportunities for fiber. End-to-end fiber-based local loops are a prospect for the future. On the other hand, feeder lines are becoming the prime areas for introduction of fiber at this time, particularly because this will facilitate the introduction of broadband ISDN, and also narrowband ISDN to an extent. About three-quarters of all new fibers installation projects are now in the feeder plant. Numerically, the feeder market is larger than the long-haul (about 45% of the total by circuit mile). Eventually, fiber will also be deployed in the distribution plant, for a general customer, providing end-to-end fiber connectivity. At this time, many commercial customers are already being equipped with fiber local loops-the large office buildings, computer centers, etc. The deployment of fiber in the local loop (distribution plant portion) for small business and residential customers will probably have to wait until the mid-1990s, though numerous experimental trials have already been undertaken. Residential fiber will open up opportunities for new services, including video-on-demand, high-definition television

(HDTV), and other graphics services (HDTV provides large-screen projection with 35-mm slide quality).

As for advanced local loops, some trends can perhaps be extrapolated from the French experiment at Biarritz in the mid-1980s. The Biarritz city test bed offered 5000 homes a host of "futuristic" services on the all-fiber network, in addition to cable TV. These services were videophone (two-way video), sensing of facilities for remote home management, video databanks (movie libraries), HDTV, and computer services (the last three were originally scheduled for a future time). Initially, fiber loops for 1.4 million homes in France were to be installed by the end of 1987, and 7 million homes were to equipped by 1992 (estimated project cost: $10 billion). The plant was to be used for multiservice broadband applications, notably cable TV (which at the beginning of the trial period in 1982 was nonexistent in France). Instead of the traditional branch-and-tree architecture, which is typical of one-way cable TV distribution applications, the plan called for a star configuration to allow full-function two-way interactive services. Because of political, technical, cost, and user-acceptance problems, that target has now been scaled down: only half million homes will be completed by the early phase. The fiber demultiplexing equipment turned out to be more expensive than anticipated. Only 30% of the houses passed subscribed to the service; videophone in particular has not been very successful. The government promoters of the system are reevaluating the cost-effectiveness of the entire concept, and of fiber in particular. Hybrid coaxial systems are now being considered in an effort to reduce cost.

2.3.1.4 Switching Systems

From a user's perspective, the primary function of a voice telephone switch is to connect, on demand, telephone instruments or other properly configured CPE. Because it would be impossible (and unrealistically expensive) to have a direct line between every possible user pairing, switching systems were developed to achieve the needed interconnectivity at an affordable price. This switched network allows a user to connect with any other subscriber by dialing the subscriber's address, and it eliminates the need for point-to-point wiring. An important aspect of any CO is the dc voltage supply that is applied to all loops. No current flows in any of the loops until the switch hook contacts of a telephone set are closed. The dc voltage permits the telephone set to signal the CO by merely closing or opening contacts with the switch hook or rotary dial.

Switches are connected by interoffice trunks and must communicate with one another to set up an interoffice call. Originally, this conversation was undertaken in-hand, namely, within the same channel as the user's conversation; however, this process was prone to fraud and was slow and inefficient. Now many switches can communicate with one another in an out-of-band fashion using separate supervisory network called common channel signaling. During the 1990s, the deployment of this signaling network will become more extensive.

Modern switches contain a common control section, which manages the call connection process, and a switching matrix, which actually makes the connection possible. Four generations of switching systems have occurred: (1) step-by-step technology, (2) common control switching systems, (3) analog electronic switching systems, and (4) digital switching systems. Modern switches are in reality computers: stored program control (SPC) implies the ability to program the switch using software instead of having to add discrete hardware modules (as was the case in older switches). Digital switches are becoming prevalent, although many analog switches are still embedded in the telephone network.

A network may contain a hierarchy of switches, beginning with local switching systems, which are closest to the user, and then going higher via tandem switches to a regional or national switch. A five-level hierarchical infrastructure was used in the United States prior to divestiture, but other topologies have now emerged.

(a) Common Control Offices

Early switches dedicated to the facilities required to control a call to each call for the duration of the call. This was inefficient and also uneconomical; if a call was blocked by equipment in the use, the system was incapable of rerouting the call. A control method that eliminated these problems was required. Equipment introduced after 1940 makes use of a limited number of specialized shared equipment units to control the process. Much of the control function to direct the path that the call takes through the system is concentrated in a small number of pieces of equipment, and these are used repeatedly. A unit of control equipment performs its function on a call and then becomes immediately available to perform the same function on another call. This mode of operation is known as common control.

(b) Computer-Controlled Switching Systems

The common control hardware can be electromechanical or hard-wired electronic (as was the case until the early 1960s) or can be computerized (which was introduced in 1965). Beginning in the late 1950s, designers realized that if the hard-wired common control was replaced with a programmable computer, options could be supplied to users more easily. Additionally, new services could be provided at the switch. As mentioned earlier, these systems are referred to as stored program control. With SPC, the control of switching functions is achieved by instructions stored in a memory and new service features may be added by changing the contents of the machine's memory. Examples of such services include speed calling, three-way calling, call waiting, call forwarding, and others.

The advantages of computer SPC include:

1. Labor saving as a result of simpler administration (for the changing subscribers' information. And reduced maintenance effort;
2. Higher traffic capacity;
3. Space saving: we can replace an existing switch with a smaller one having a larger capacity;
4. Power saving;
5. Cost reductions due to continued VLSI cost improvements;
6. Flexibility to changes over the life cycle of the switch (which could be as long as 40 years); and
7. Economical offering of new advanced services to the subscriber.

The Bell system's first SPC switch was the number 1 Electronic Switching System (ESS), which went into service in 1965. Number 1 ESS was designed to handle the heavy traffic loads and high density of telephone customers in metropolitan areas. The same basic principles were used in the No. 2 ESS, introduced in 1970. The No. 2 ESS was intended for communities with local switching offices serving around 10,000 telephone lines. Three basic elements of this type of switch in its early manifestation were: (1) a switching matrix using high-speed electromechanical ferrite switches; (2) a control unit, which directs the operations and maintenance in the system; and (3) two memories: a temporary memory (call store) for storing information such as the availability of circuits, called number, calling number, and type of call; and a semi permanent memory (program store) containing all the information that the control unit needs to process the call and make a connection. The program store is semi permanent because it does not have to be changed as calls are processed by the system. Two control units and maintenance frame make up the control complex; switches are designed with two control units with lock-step full duplex processing so that no service degradation is experienced for active calls (or calls being processed) should a single processor fail. The various subunits that form a control unit include the program control, input-output control (of peripheral units such as switching networks), the call store, and the program store.

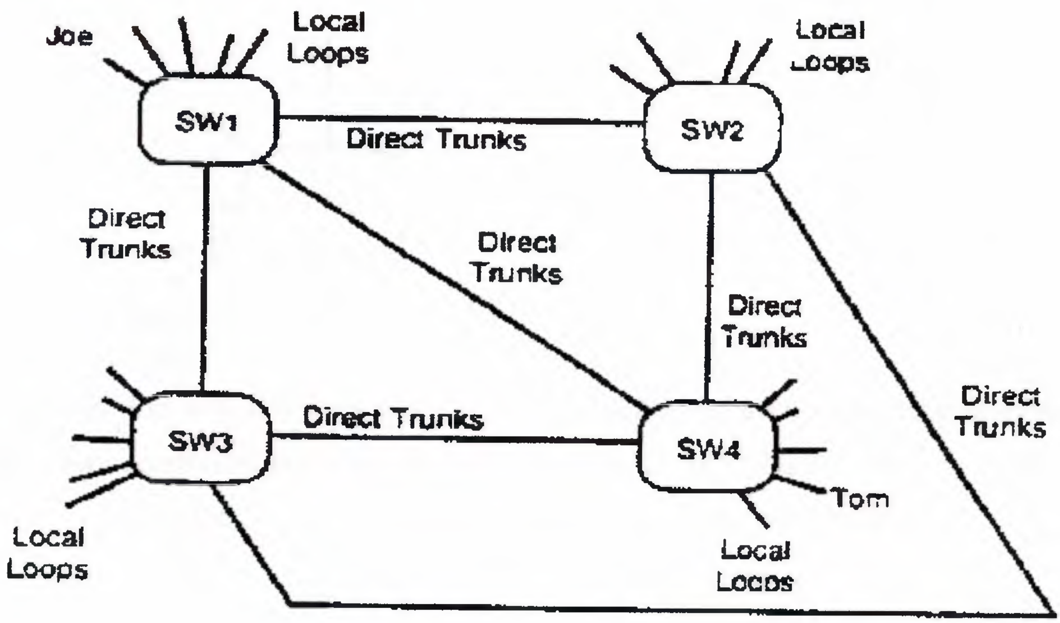
(c) Digital Switching Offices

Initially, switching was achieved by activating electromagnetic relays that would close to achieve a continuous metallic circuit end-to-end. This type of switching, also called analog, has a number of drawbacks, including unreliability due to mechanical components, noise added to signals due to the opening and closing of relays, large size inconsistency with digital transmission systems, and other drawbacks. Digital switching technology accepts digital signals and switches these to the desired destination by redistributing the signal electronically.

Voice digitization techniques have been widely applied, as indicated earlier, and their use is growing. Digitization is valuable in transmission because of its ability to protect the signal against the corruptive influences that degrade analog transmission. The use of corresponding digital carrier systems is increasing at such a rate (more recently because of the use of fiber) that to switch the digital signal directly is more advantageous than converting it to analog for space-division switching, then re-encoding it for transmission.

(d) Interoffice Trunks

Telephone switches are interconnected by a group of circuits called interoffice trunks. Typically these trunks are pooled, meaning that they can be seized by switches at either end and put on line as needed. The size of the pool is calculated in such a fashion that at the busy hour in the busy season no more than 1% (or some other small number) of the calls are unable to be routed to the next stage of the switching over the trunk pool system. Trunks are usually derived from carrier system of trunks, they are said to have direct trunks between them. (See Figure 2.6).



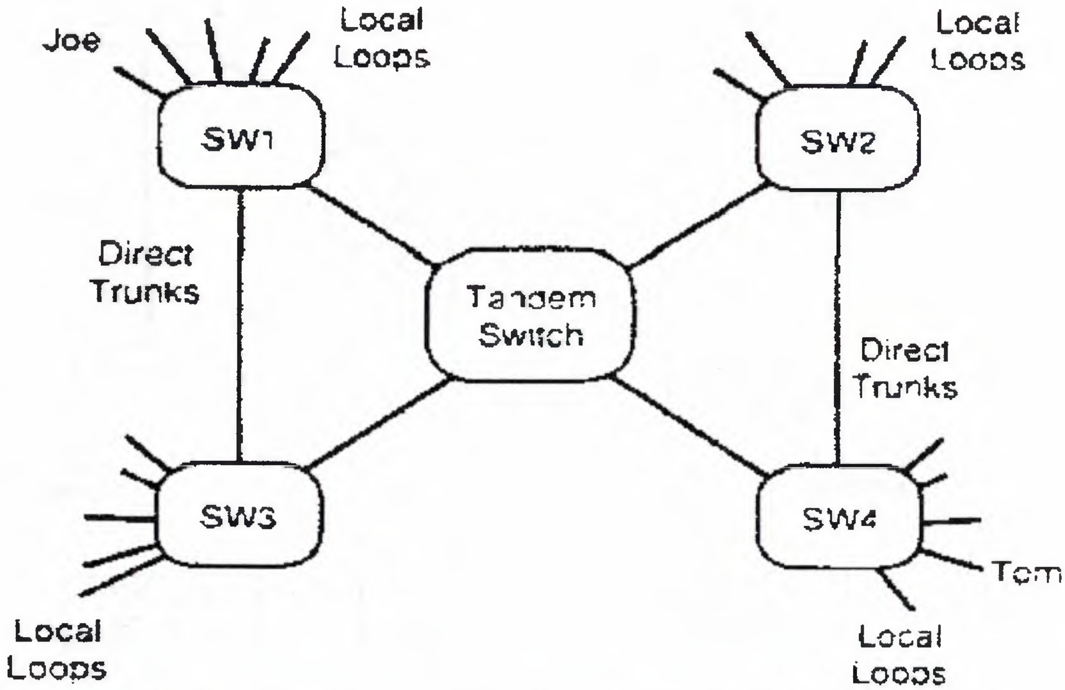
All witches are connected with trunks.

Joe talks to Tom over the trunks system between SW1 an SW4.

Figure 2.6 Direct Trunks.

(e) Tandem Switching Offices

In telephone networks with a large number of COs, having trunks between every pair of switches would not be practical. In these cases, the local telephone exchanges are connected to trunks that can provide access to an intermediate switching center, known as tandem switching offices, to which all switches are in turn connected. Connectivity between two local COs that are not directly connected by direct trunks must go through the tandem switch. Today, access to toll facilities is typically (but not always) via an interexchange tandem. (See Figure 2.7).



SW1 and SW3 as well as SW2 and SW4 are connected with direct trunks because of high traffic. Other connectors are made via the tandem switch. Joe talks to Tom over trunks to the tandem switch, the tandem itself and then over trunks to SW4.

Figure 2.7 Tandem arrangement.

(f) Interexchange Trunks

As a product of divestiture, long-haul communication is handled in almost all situations by a carrier other than one providing the local loops and the local switching system. A tandem arrangement may be employed to reach one or more interexchange carriers. (See Figure 2.8).

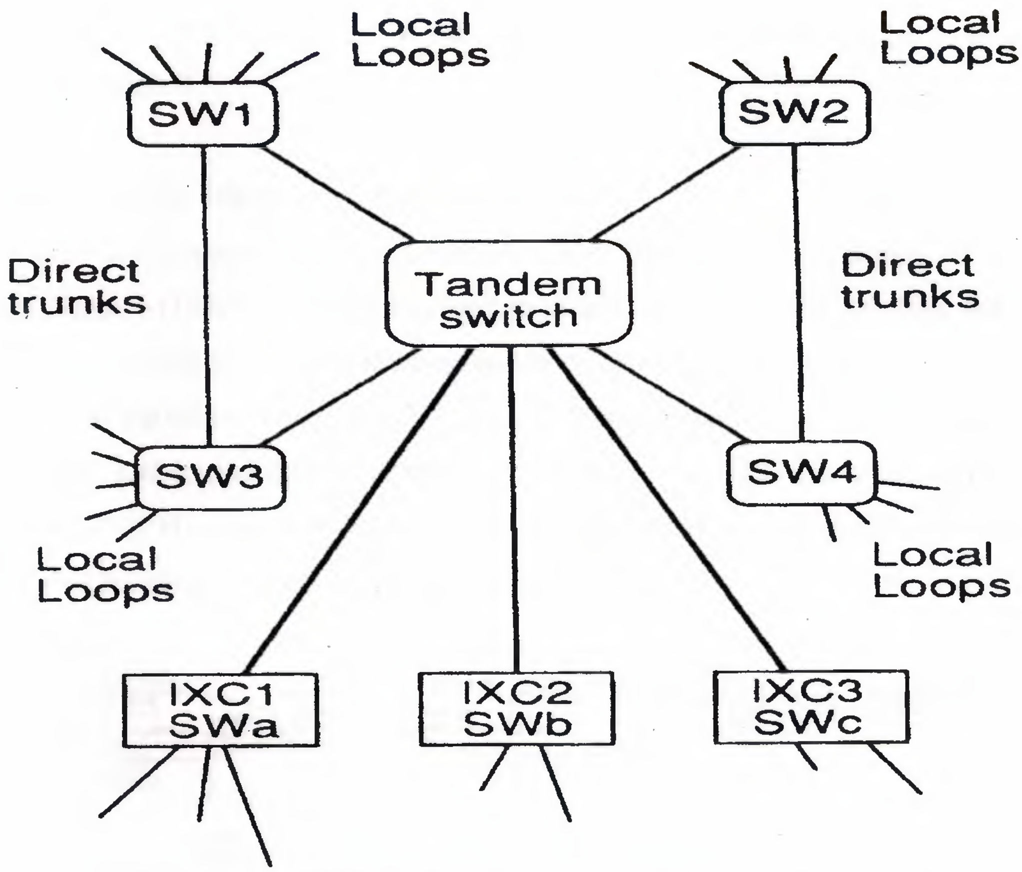


Figure 2.8 Typical access to long-haul carriers.

(g) Traditional Switching Hierarchy

A traditional tree-based hierarchy of switches ensures that there is a path from each switching office in the network to any other switching office in the network. It is an architecture, which had been employed by the Bell System for decades. A characteristic of the hierarchical structure of switching offices is that each office is connected to an office at a higher level except, of course, for those at the highest level; these top level offices are completely interconnected. The traditional network structure for the United States and Canada is divided into 12 regions; each region has one switching office at the highest level. The trunk group that connects a switching office to the next highest level switching office within a region is called a final group. Additional trunk groups supplementing the tree structure are permissible; in fact, they are desirable where sufficient traffic exists between switching offices not directly connected by the tree structure. These trunk groups, which are not part of the tree structure, are called high-usage groups. If one group is busy, a call will be rerouted to a different group until the final group is reached.

(h) Evolution of the Telephone Company Plant

Figure 2.9 depicts the five-stage evolution of the telephone company plant over the years. Stage 1 (1890s to 1950s) involved an all-analog plant. Stage 2 (1960s and early 1970s) saw the emergence of digital transmission. Stage 3 (mid-1970s and 1980s) saw the introduction of digital switching. Stage 4 (early 1990s) is experiencing the introduction of ISDN for true end-to-end digital connectivity. All these stages have involved voice band bandwidths (4000 Hz or 64,000 b/s); in stage 5 (late 1990s) we will see the introduction of end-to-end broadband digital communication.

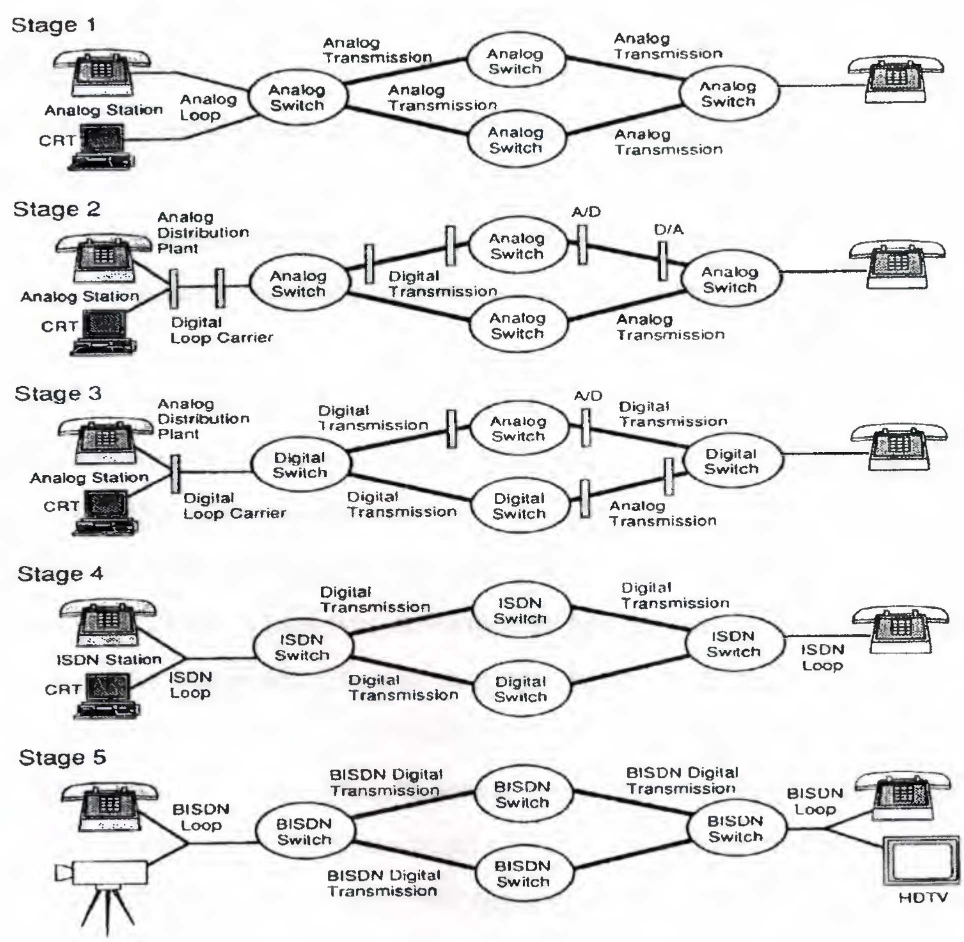


Figure 2.9 Evolution of Telco plant.

(i) North American Numbering Plan

Numbering schemes are mandatory for orderly identification of network subscribers. In traditional telecommunication networks, numbering plans are also used to accomplish routing.

A domestic toll voice call (with presubscription to an interexchange carrier) can be placed using an address such as 1-NXX-NXX-XXXX, where the first three digits after the 1 represent the area code, the middle three digits the exchange, and the last four digits the specific station connected to the switch specified by the NXX. A total of 160 area code combinations is possible under the current numbering plan, which requires the first digit to be a number between 2 and 9 (an "N"), the second digit to be a 0 or a 1, and the third digit to be between 0 and 9 (an "N"). code combinations 211,311,411,511,611,711,811,and 911, as well as 200,300,400,500,600,700,800, and 900 are reversed for special applications. When AT&T first assigned the initial area codes for the United States, Canada, and parts of the Caribbean in 1947, engineers projected that they would last 100 years. With a potential one billion phone numbers now assigned, the 100-year projection will fall short by more than 50 years. All currently available area codes will be assigned by 1995. The overall growth in telephone number usage is running at about 7 to 9% a year. The growth is fueled by growth in population, multiloop households, cellular telephony, paging, and facsimile machines.

Two recently introduction area codes are 708 in Illinois and 908 in New Jersey. Area codes that have been reserved include 903 (Dallas, late 1990); 510 (San Francisco, late 1991); and 310 (Los Angeles, a992). The remaining unassigned codes under the original numbering plans are: 210, 410, 706, 810, 905, 909, 910, and 917.

During the 1960s, a plan was developed to provide for the time when the existing area codes would be exhausted. The plan removes the requirement that the second digit of each area code be a 0 or a 1; eliminating this restriction makes 640 additional codes available. Implementation of the plan, however, is a major undertaking for all LECs. Each of the 792 COs in area code has a maximum of 10,000 telephone numbers associated with it, which means that each area code has approximately 7.9 million numbers than can potentially be assigned to customers.

(j) Traffic Engineering

The traffic offered to a switch is a function of two factors: the average rate of arrival of new call attempts and the average holding time of a call (assuming that the variance is small enough to be safely ignored). The averaging period for the origination rate is the busy hour, a one-hour period chosen to typify for a given CO the annually recurring hour during which the offered traffic load is a maximum. Peak busy hour calls is the unit used for expressing the processing capacity of a switching machine's control. The offered traffic load is expressed in hundred call seconds ("CCS" where the first C is from centum = hundred), and is the product of the number of calls and the average holding time, or the sum of the holding times of calls under consideration. By convention, the units of CCS are often used to mean CCS per hour. The average holding time multiplied by the number of calls placed per unit time is a measure of the traffic intensity and is often expressed in erlang, named after an early contributor to traffic theory, is the traffic intensity equivalent to one call held for an entire hour, and is therefore equal to 36 CCS per hour (it is also equivalent, for example, to 36 calls held for 100 seconds each in a one-hour period, or a circuit occupied 100% for a full hour by any number of calls). Put differently, the traffic intensity in erlangs is the number of channels that would be sufficient to serve a given offered load in a one-hour period, if the load timing could be rearranged so that all the channels would be continuously busy. It thus constitutes a lower bound on the number of channels to carry that traffic intensity.

2.3.2 Data Networks

While data networks differ in many ways from voice networks, the components identified above (CPE, switching, and transmission) are all still present. CPE may consist of CRTs (cathode ray tubes), computers, remote peripherals, *et cetera*. High-speed private networks, such as LANs, usually do not include an explicit tance or rejection based on the address posted on the message (the message is broadcast to all users of the network). Lower speed networks may involve circuit-switched or packet-switched facilities in the network, or switching in the host computer or associated front-end hardware.

2.3.2.1 CPE

CPE in the data environment includes:

- CRTs, printers, plotters, computers;
- Local area network (LANs);
- Private wide-area networks (WANs);
- Modems, multiplexers, channel service units; and
- Front-end processor.

(a) DTE and DCE

A data communication system consist of data terminal equipment (DTE), data circuit-terminating equipment (DCE)-colloquially known as data communication equipment-and the transmission circuit, also variously known as channel, line, link, or trunk. The DTE is a device, such as a terminal or a computer (mainframe, minicomputer, or microcomputer). The DTE supports end-user applications, for example, data entry, inquiry or response, and database management functions. The DCE provides the connection of the user DTE into the communication circuit. Notice that both DTE and DCE can be CPE (although some DCE may also be part of a public network). (See Figure 2.10).

A physical level specification (also know as a physical level interface), such as EIA 232-c, defines the following attributes of a data communication system:

1. The wiring connection between devices (when wires are used);
2. The electrical, electromagnetic, or optical characteristics of the signal between communicating devices;
3. The provision for mechanical connectors (dimension, number of pins, etc.);
4. The agreement on the type of clocking signals that will enable the devices to synchronize onto each other's signal;
5. The provision for electrical grounding (if needed).

For example, RS-232-c describes a standardized interface between DTE and DCE employing serial binary data interchange. RS-449 describes a general-purpose 37-position and 9-position interface for DTE and DCE employing serial binary data interchange.

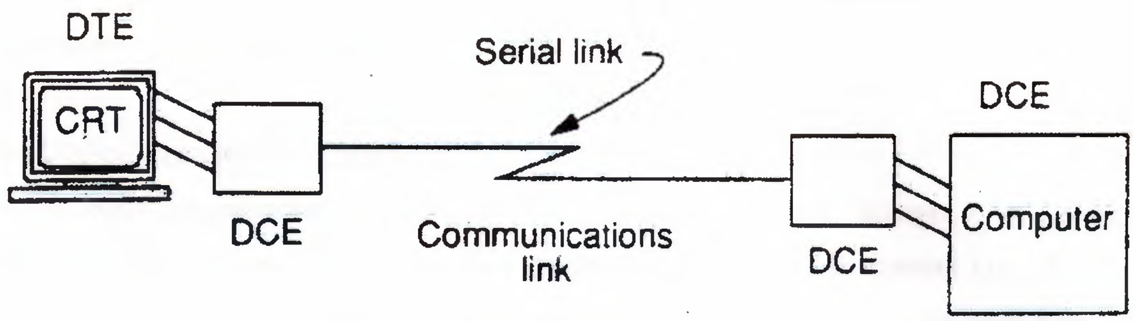


Figure 2.10 DTE and DCE.

(b) ASCII and EBCDIC Character Coding

One of the issues in data communication is the representation of the data, particularly when dealing with several computer systems. The applications in different systems may wish to represent data structures in different ways. Yet a common exchange structure is needed. Many computers use the 7-bit ASCII (American Standard Code for Information Interchange) code; other systems use the 8-bit EBCDIC (Extended Binary Coded Decimal Interchange code).

2.3.2.2 Transmission

The transmission link can be analog or digital. It can also be point-to-point or multipoint. Additionally, the link can also be characterized as being two-wire or four-wire, and half-duplex or full-duplex. These concepts are summarized below.

(a) Analog Transmission Methods

Analog signals vary in time in terms of amplitude and frequency. As indicated earlier, at the current time, analog network interfaces are still very common. To send data over an analog line we use a modem. The modem takes a digital signal and produces a signal suitable for transmission over the analog network.

The analog signal generated by a modem to transmit data consist of a carrier frequency, plus sidebands that change as the data bit pattern varies. These sidebands must fit within the attenuation limits of the voice-grade channel. If the timing will be distorted when reproduced by the distant modem. To minimize the effects of channel attenuation, the modems may include or require conditioning filters to produce an attenuation curve which complements that of the channel.

(b) Digital Transmission Methods

New transmission facilities, specially designed for data, accepts a digital signal directly, although a network termination device is still required to connect the CPE to the public network. The internal transmission method, however, is still based on analog media.

(c) Types Of Circuits

Point-to-point and Multipoint Data Circuits. A data circuit can also be classified as point-to-point or multipoint. A point-to-point circuit connects two devices only and a multipoint circuit connects more than two devices.

Two-wire versus Four-wire. The communication channel on the line side of the DCE is usually described as "two-wire" or "four-wire." These terms are derived from copper-based telephone nomenclature, in which two or four wires (or equivalent) are employed to transmit information. Four-wire affords more flexibility, but requires more transmission resources (i.e., cable). Typically, one pair of wires is used for transmission of information in each direction, and higher effective bandwidth is achievable. With two wires, more sophisticated electronics is required to achieve intelligible simultaneous two-way data transmission. The majority of traditional local loops for voice use employ two wires to minimize the investment in copper, as discussed earlier. Long-distance voice circuits employ four wires, so that a separated transmission path is used for each direction. The interface between the two configurations is provided by a device called a "hybrid."

Data circuits that need to carry 9.6 kb/s or higher on a sustained basis have traditionally consisted of dedicated transmission facilities (also known as private lines), which are four-wire equivalent end-to-end. Dedicated four-wire circuits are particularly common in multipoint applications. Two-wire, Full-duplex. Echo-canceling, error-correcting, high-throughput modems for use with switched lines are increasingly becoming available.

Half-duplex versus Full-duplex. These terms describe how data are transmitted across the channel, regardless of the physical configuration of the channel. Half-duplex defines a transmission in which data are transmitted in both directions but not at the same time. The DCE alternates between transmitting and receiving the data between the devices. Full-duplex describes the simultaneous transmission of data in both directions. The term "two-way alternate" is also used to describe a half-duplex data flow and "two-way simultaneous" to describe a full-duplex data flow.

The ability to operate in a full-duplex mode depends on: (1) the channel (i.e., if the echo suppressor found in a long-haul dialup circuit can be disengaged); (2) the modem; and (3) the protocol used by the communicating entities. If any one of these three components fails to operate in a full-duplex mode, then the overall transmission becomes (effectively) half-duplex. Today, the majority of the data communication systems are full-duplex.

(d) Line Turn-Around

When the DCEs use half-duplex schemes, a time interval is required for the devices to stabilize and adjust to the signal before transmission in the other direction. This stabilization period is called "training time," and the process of reversing the signal is called "line turn-around". Turn-around times of 100 to 200 ms are not uncommon. Because of this delay, most multipoint systems keep the master device's carrier on constantly (constant carrier) with the slaves configured for switched carrier operation (switched carrier). This approach eliminates half the turn-around delay. Some DCEs have used split channel techniques to eliminate the switched carrier delay completely.

(e) Asynchronous and Synchronous Transmission

Two methods for formatting and transmitting user data through the channel exist: asynchronous and synchronous. The asynchronous approach is an older technique; however, due to its simplicity and relative low cost, it is still a common method and is found in many modern systems such as personal computers. Asynchronous transmission is characterized by the use of timing bits (start and stop bits) enveloping each transmitted character. The purpose of the start and stop bits is to provide for character synchronization and timing between the transmitting device and the receiving device. The stop bit notifies the receiving device that a character is being transmitted on the channel. The stop bit indicates that all the bits have arrived and provides for each other's timing functions. In asynchronous transmission, each character is framed by start and stop bits.

In the synchronous approach, all characters are directly blocked together and are transmitted without the intervening start and stop bits. Framing codes (called syncs or flags) are placed in front and behind the full data unit (usually called a frame) to indicate to the receiver where the user data begins and ends. Synchronous transmission can provide timing signals by one three techniques:

- A separate clocking line. A separate clocking line is a technique used for short-distance nontelecommunication connections: in addition to the data line, another line transmits an associated timing signal, which is used to clock the data into the receiver. For example, the EIA-232-D and CCITT V.24 specifications provide several options for synchronous transmission and clocking. Separate clocking channels are not practical for longer distances because the installation of a separate link or wire is expensive. Longer distances also increase the probability that the clocking line will lose the synchronization with the data line because each line has its own unique transmission characteristics. The telephone network does not provide clocking lines.
- Embedding the clock signal in the data stream with the data acting as a clock to a simple receiver circuit. To embed the clocking signal, the data bits are encoded at the transmitter to provide frequent transitions of the channel.
- Embedding the clocking signal in the data stream and using it to synchronize a receiver clock. The line transitions in the incoming bit stream keep the receiver clock aligned (synchronized) onto each bit in the data block.

(f) Digital Modulation

The principle of digital modulation are the same as those of analog modulation discussed earlier, except that the modulating signal is digital. In a digital environment, the modulator accepts binary encoded symbols and produces waveforms appropriate to the physical transmission medium at hand, which is always analog with today's technology. Because these signals involve discrete jumps from one state to another, the modulating action is generally described as keying.

The simplest technique, amplitude-shift keying (ASK), modulates the carrier with the binary signal to produce an AM signal. Binary 0 is represented by no amplitude of the carrier and binary 1 with full amplitude of the carrier. The off-on keying is simple, but ASK makes inefficient use of transmission power. Amplitude modulation is not often used by itself because of transmission line power problems and sensitivity to line errors. However, it is commonly used with phase modulation to yield a method superior to either FM or AM.

Frequency-shift keying (FSK) is also common. This simple FM technique uses the binary signal to switched between two frequencies. In its simplest form, it is characterized by abrupt phase changes at the transition between one state and the other, which distorts the spectral energy and reduces spectral efficiency.

Phase-shift keying (PSK) is also being increasingly used because of higher efficiencies achievable with this technique (packing more data onto the carrier, or "more bits per baud," in engineering jargon). Simple PSK uses the binary signal to alternate the phase of carrier between 0 degrees and 180 degrees. A variation, known as differential PSK, compares the phase of the previous bit with the present bit to determine if a transition has occurred.

Efficiencies of around 1 b/Hz of bandwidth are achieved with basic PSK, FSK, and ASK. Two bits per hertz can be obtained through the use of 4-phase PSK and quadrature amplitude modulation (QAM) (QAM is discussed below). More complex schemes, such as 16-phase PSK and 16-state QAM, are needed for 3 b/s per hertz. Four bits per hertz is achieved with 64 QAM. However, as the complexity of the modulation techniques increases, a better S/N is required to achieve the same bit error rate (BER) grade of service. (To be more precise, we mean bits per baud, which are changes in Signal State; for example, the V.32 modem operating at 9.6 kb/s has a modulation rate of 2400 baud and encodes 4 bits/baud).

(g) The Modem

As indicated earlier, the modem is responsible for providing the required translation and interface between the digital environment and an analog communication link. Modems are DCEs, and have become CPE in the United States. The modem provides a digital-domain to voice-domain conversion; modems should not be confused with the analog-to-digital process discussed earlier under the PCM technique. Modems are designed around the use of an analog carrier frequency in the voice band domain. The carrier is modulated with the DTE's data stream. The carrier signal is changed back to digital at the receiver DCE modem by the process of demodulation. Modems can use amplitude, frequency, and phase modulation, or a combination thereof to impart the data signal over the analog carrier. Each method impresses the data on a carrier signal, which is altered to carry the properties of the digital data stream.

Amplitude modulation modems alter the carrier signal in accordance with the modulating digital bit stream, while the frequency and phase of the carrier are held constant (the carrier is turned on or off, or at least changed in magnitude).

Frequency modulation modems alter the frequency of the carrier in accordance with the digital bit stream. The amplitude is held constant. In its simplest form, a binary 1 is represented by a specified frequencies and a binary 0 by another. The most common FSK modems use four frequencies within the useable bandwidth of a standard telephone circuit (nominally, the bandwidth is 4000 Hz). A typical full-duplex two-wire FSK modem transmits 1070-and 1270-Hz signals to represent a binary 0 (space) and binary 1 (mark), respectively. It receives 2025- and 2225- Hz signals as a binary 0 and binary 1. FSK has been a widely used technique for low-speed modem (up to 1200 b/s). It is relatively inexpensive and simple; many PCs use FSK modems. The most sophisticated modems employ multiple carriers, instead of two carriers.

Phase modulation modems alter the phase of the signal to represent a 1 or 0. A common approach to implement PSK is to compare the phase of the current signal state to the previous signal state. PSK techniques use bandwidth more efficiently than FSK, but they require more elaborate equipment for signal generation and data representation. Phase modulation techniques are almost exclusively used today on high-speed modems. Phase shifting can be used to provide multilevel modulation. Table 2.1 depicts a 4-phase PSK and an 8-phase PSK modulation scheme. In the former we can map 2 pits per baud (one 360-degree carrier signal rotation); in the latter, 3 bits per baud.

An extension of multiphase PSK modulation is quadrature amplitude modulation (QAM), in which the sine and cosine of the modem carrier frequency are amplitude modulated with two or more amplitudes. QAM techniques are widely used in high-speed modems. QAM is a combination of PSK with an amplitude for a set signal points.

Most modems use scrambling techniques to ensure a proper number of state transitions for accurate timing recovery at the receiver modem. The scrambling is usually done by the DCEs. Scrambling modem (DCE-to-DCE) synchronization. However, synchronization of synchronous transmission between user devices and modems (DTE-to-DCE) must be performed with a separate timing (or clocking) circuit.

Most modems that operate with speeds up to 4.8 kb/s employ fixed equalizers; these circuits are designed to compensate for the average conditions on a circuit. However, the fixed equalizers are being replaced with dynamic (or automatic) equalization: the modem analyzes the line conditions and adjusts its equalization accordingly. The adjustments occur very rapidly, on the order of 2400 times a second for a 9.6 kb/s modem. More information on equalization follows.

Table 2.1 Quadrature Modulations

4-PSK	8-PSK		
Bits to Code	Phase Change in Signal (deg)	Bits to Code	Phase Change in Signal (deg)
11	45°	111	22.5°
10	135°	110	67.5°
01	225°	101	112.5°
00	315°	100	157.5°
		011	202.5°
		010	247.5°
		001	292.5°
		000	337.5°

(h) History of Modems

The 70-year history of voice band modems falls four into phases:

1. From 1919 to the mid-1950s, work arose out of the need to transmit telegraphic information over the voice network. Research focused primarily on the basic properties of copper lines and on basic theories of data communication. The maximum data rate was around 100 b/s.
2. Starting in the mid-1950s, growing military requirements, and nascent commercial interest in transmitting large amounts of data, led to efforts to achieve greater transmission speeds. The technical investigation concentrated on modulation techniques, telephone line characteristics, and receiver design. Methods to deal with marginal phase distortion in additive noise channels, equalization, intersymbol interference, channel amplitude distortion, and delay distortion were developed. This resulted in an increase in the speed from 100 b/s at beginning of this period to 9600 b/s in the late 1960s. By the late 1950s, AT&T introduced the bell 103 (300 b/s) and bell 202 (1200 b/s) modems; these employed FSK principles. In the early 1960s, the application of 4-phase PSK modulation, resulted in the bell 201, which provided 2400 b/s over conditioned lines. Commercial products in the late 1960s provided reliable higher speed bandwidths; notable in retrospect were Milgo, which achieved 4800 b/s with 8-phase PSK, and Codex, which achieved 9600 b/s with QAM techniques with a 16-point signal constellation (i.e., 4 b/Hz).
3. During the 1970s, the speed on commercially available modems remained around 9600 b/s, but major design improvements led to significant reductions in size and power. Techniques implemented in this period, including LSI and VLSI, timing recovery, adaptive filtering, and digital signal processing, would establish the basis for the advancements of the next phase. Some improvements in speed (to 14,400 b/s) were obtained by using more advanced equalization techniques.

4. During the 1980s, error-correcting modems, advanced signal processing, and higher speeds were introduced-19,200 b/s is now routinely possible on dedicated lines. Speeds as high as 38.4 kb/s are now achieved using data compression. Because telephone lines use fiber for a larger for a large portion of their total span, they are less prone to some of the traditional problems, including phase jitter; S/N of better than 28 dB are achievable. QAM signal constellations with 64 points (6 b/Hz) become practical. The 19,200 b/s modems use orthogonal multiplexing (transmission of several noninterfering subsignals over the common channel), or multidimensional trellis-coded modulation, which we will discuss latter. In band-limited channels an increase in transmission rate requires an increase in the number of coding points in the constellation; this, however, can run into marginal-performance areas (in terms of signal quality) of the channel. An approach to dealing with possible mutilation of some constellation points is to use error-correction bits. Until the early 1980s, however, it was thought that the increased speed would wash out compared to the overhead needed to provide the needed error correction. This turns out not to be true. Trellis-coded modulation can improve the performance of a modem by 3 to 6 dB.

In spite of repeated predictions over the past decade that modems would soon be eliminated by end-to-end digital networks, the modem industry continues to prosper. While digital backbones are becoming popular, a large portion of data communication in 1990 is still carried by voice band modems over the analog telephone network. Modems are now available on one or two chips. Effective bandwidth has increased approximately 30-fold during the past decade, and the cost per bit per second has decreased 20-fold. In the early 1980s, a 1200 b/s modem cost \$1000, providing 1.2 b/s per dollar; in 1987 one code obtain 5 b/s per dollar (\$2,000 for a 9.6 kb/s V.32 modem); in 1990 one can obtain 16-18 b/s per dollar (\$2,000 for a top of the line 38.4 kb/s V.32 modem with error correction, or a 9.6 kb/s V.32 modem for \$600). Both the bandwidth and the cost per b/s have improved for the dialup modems as well as for the private line modems. Table 2.2 provides an assessment of both the actual growth of data, and of digital facilities in particular, and may be used to approximate the total data rate in the United States. The long-haul data carried on analog private lines in 1989 is estimated to be 3 billion b/s; that number decreases to 0.3 billion b/s by 1993. (Remember, however, that a lot of data are and will continue to be carried on dial-up facilities). The long-haul data carried on digital private lines in 1989 are estimated to be 23 billion b/s; that number increases to 120 billion b/s by 1993.

Table 2.2 Demands for Telecommunications Services

Transmission Service	Number of Private Lines			
	1989 IXC[f]	1989 LEC[g]	1993 IXC[f]	1993 LEC[g]
Analog dedicated circuit	303000	1035000	36000	100000 [c]
Digital dedicated circuits	47000[a]	88000	310000[b]	600000 [c]
Fractional T1 circuits	1600	(d)	21000	(e)
DS1 circuits	82000	65000	16000	110000 [c]
DS3 circuits	200	2000[c]	1200	10000[c]

a 50% are 56 kb/s lines.

b 90% are 64 kb/s lines.

c Estimated.

d not generally tariffed.

e Unknown if tariffed.

f Interexchange Carrier.

g Local Exchange Carrier.

(i) Compatibility Issues Pertaining to Modems

Given the plethora of techniques available to the modem to encode the digital signal for transmission over an analog circuit, we need to ascertain that the modems at both ends of the circuit are compatible, preferably following some established standard. In theory, two modems designed to the same standard should interoperate. In practice, people generally bay the two modems needed on a link from the same vender; this is particularly true for higher speed modems that maybe using proprietary encoding, modulation, error correction, and data compression schemes.

CCITT has published a series of recommendations to bring some standardization to the equipment. For example, CCITT Recommendation V.22 describes standardized 1200 b/s full-duplex modems for use on the general switched telephone network. V.29 describes standardized 9600 b/s modems for use on point-to-point leased circuits. Table 2.3 depicts some key modem families.

In 1982 CCITT started to develop recommendations for full-duplex two-wire modems operating at 9600 b/s over the public switched network. This was the first time that a modem standard was developed prior to a commercial product; in the past, all modem recommendations had been developed based on successful commercial products. An eight-dimensional error-correcting code is included in recommendation V.32 (two-wire switched line, 9600 b/s) and in V.33 (4-wire dedicated line, 14,400 b/s). V.32 bits (possibly a standard by the end of 1991) is designed to let a V.32-standard 9600 b/s modem operate at 14.4, 12.0, 9.6, 7.2, and 4.8 kb/s in full duplex over a dialup link, with on-line rate negotiation.

The new V.42 standard, primarily concerned with error correction, was formally adopted by CCITT late in 1988. The protocol includes both Microcosm's Networking Protocol (MNP) and CCITT's LAPM (link access procedure for modems). LAPM uses cyclic redundancy checking CRC) to detect errors in transmission and recover with a retransmission. Because an installed base of a half million error-correcting modems using MNP already exists, V.42 included both options. MNP is a de facto standard for error correction that has evolved through several "classes". The basic classes provide error detection and correction; the more advanced classes include compression techniques. Microcom released the first four classes of MNP to the industry and they license Class 5 and Class 6 to vendors that desire to include the protocol in their own products (the higher classes were still proprietary in 1990). Currently, only a couple of vendors have a V.42 product.

Table 2.3 Some Key Families of Modems

Modem Type	Special and Characteristic	1990 Price
Bell 103A	300 b/s full-duplex operation on dial-up line	$20-50
Bell 212A	1200 b/s full-duplex operation on dial-up line	$50-100
V.22 bits	2400, 1200 b/s full-duplex operation on dial-up line	$100
V.22 bits	19,200 b/s full-duplex operation on dial-up line, with compression	$700
V.29	9600, 4800 b/s half-duplex (2-wire) and full-duplex (4-wire) leased line	$800
V.32	9600, 4800 b/s full-duplex operation dial-up line with echo cancellation	$600
V.32	9.6 kb/s to 28.8 kb/s to full-duplex operation on dial-up line with MNP error correction	$1200-2000
V.42	38.4 kb/s-19.2 kb/s with error correction and compression	$1500

V.42 bits aims at providing data compression. Compression ratios of 2-to-1, or even 4-to-1 may be possible. This means that a 9.6 kb/s modem can provide an effective throughput of 38.4 kb/s. the compression is based on an algorithm known as Lempel-Ziv. As of early 1990, only one modem on the market in the United States utilized this technique (the Telebit Trailblazer ™ modem, which operates at 19.2 kb/s, first used an earlier version of a compression algorithm, but now is fully compliant with V.42 bits). Formal approval of the V.42 bits was planned for 1990.

The multitude of modem standards often leads to interoperability problems. Many modem manufacturers now include multiple transmission schemes into a single modem. Such modems allow users to choose an appropriate scheme for the application at hand, and facilitate interoperation with various existing modems. As of early 1990, more than half a dozen companies provided these "multimodems". One such product can, for example, implement nine distinct modulation techniques.

(j) Capacity of a Data Link

The maximum digital capacity of an analog communication channel, in b/s, is given by Shannon's equation:

$$C = W \log_2 (1 + S/N)$$

Where

C = channel capacity in b/s,

W = channel bandwidth in hertz,

S = signal power in watts,

N = noise power in watts.

[If the S/N is expressed as x decibels, then $S/N = 10^{(x/10)}$].

Application of Shannon's equation to a voice-grade line with a S/N of 30 dB (which is typical given the amount of power that can be applied to a copper medium, and effects of noise and impairments-39 dB being the theoretical maximum) leads to a maximum bandwidth of 30,000 b/s for a 3000-Hz telephonic voice-grade channel [obtained as 3000 $\log_2 (1+10^3)$]. When a modem has effective throughput of 38.4 kb/s, it is achieved with data compression techniques along with efficient modulation schemes (producing perhaps an uncompressed throughput of 19,200 b/s; the 2-to-1 compression algorithm would produce an effective rate of 38.4 kb/s). This does not contradict Shannon's equation. A higher data rate could also be carried over an unloaded copper wire with no filters because the bandwidth W is higher. A 22-gauge copper wire can, in theory, carry up to 5 MHz, allowing higher digital throughput, as, for example, in the case of twisted-pair LANs, digital loop carrier systems, and ISDN local loops.

(k) Source of Noise in Data Circuits

Many physical factors inhibit successful data transmission. Some of the factors applicable to a traditional telephone channel are Attenuation, Cross talk, Thermal noise, etc., this discussion should clarify the need for digital networks, fiber-based networks, and networks designed specifically for data communication.

1) Attenuation

Attenuation (A) is loss of energy in the signal as it is transmitted through the medium. It is described in decibels and is defined as:

$$A = -10 \log_{10} P_{in}/P_{out} \text{ dB}$$

Where P_{in} is the input power at one end of the channel and P_{out} is the output power at the other end of the channel. Without amplification, the power out will be less than the power in. for example, a loss of 50% of the power corresponds to a 3 dB loss. A gain of 3 dB is a doubling of power. The measure "dBW" is often used to indicate the gain relative to 1 W of base power. Attenuation affects all types of transmission systems including coaxial-based systems, microwave systems, and fiber optic systems.

In a band-limited channel, such as a traditional long-distance voice circuit, designers customarily design for minimum attenuation in the middle of the spectrum. The loss is usually measured with reference to the 1000-Hz level. The useful bandwidth of a copper circuit is expressed as the frequency difference between points on the attenuation curve that represent 10 dB of loss with respect to the level of a 1000-Hz reference signal.

Attenuation versus signal level refers to dynamic loss in the channel, such as that caused by a compander operating at a syllabic rate. A compander is a device installed in the voice channel that amplifies weak signals more than strong signals at the source, and amplifies strong signals more than weak signals at the destination, to restore the original characteristics. It is fast acting, following the syllables of speech for maximum effect. The companding technique permits the voice signal to be transmitted at a higher average level than would otherwise be the case and results in a consequent improvement of the S/N at the receiver. In data transmission, companding has the undesirable effects of decreasing the apparent modulation percentage while producing spurious frequencies that show up as noise. Line conditioning can ameliorate the situation.

2) Cross talk

Inductive, capacitive, or conductive coupling between adjacent circuits results in having an undesired signal "leak" across and appear as a weak background signal added to the desired signal. Cross talk degrades the S/N and increases the error rate. A limit is placed on the amount of noise that is allowed on the circuit. If the noise can not be attenuated nor than 24 dB below the signal, the circuit may need to be repaired or changed.

3) Echo

When an electrical circuit such as a telephone line is not properly terminated, some of the data signal energy is absorbed by the receiver and some energy, called echo, is returned to the sender. A portion of the echo may be reflected by an improper line termination at the sender and reach the receiver as a noise. Like other kinds of noise, it increases the error rate, unless the signal level of the data is much greater than the signal level of the echo. The ability of a circuit to reduce reflections is called echo return loss and is measured in dB. The greater the return loss, the better is the termination. A ratio of 24 dB or more is desired.

4) Transients

Transients include impulse noise, gain hits, phase hits, and dropouts. Impulse noise is defined as any noise that exceeds the root mean squared level of the background noise by 12 dB for more than 10 ms. This kind of noise can be expected on most metallic facilities. It is the result of coupling from nearby circuits that carry electrical current surges originating in switching equipment, line faults, or from lightning surges. The desired data signal usually exceeds the background noise by at least 26 dB. A correlation exists between the frequency of occurrence of noise impulses and the number of electromechanical switching offices through which the data circuit must pass. Reduction of these switches in the 1980s has had the effect of decreasing the importance of switches as a source impulse noise.

Telephone circuits are rated according to the average number of "hits" of impulse noise that exceed threshold over a period of time. When the statistical average is exceeded, an attempt ought to be made to find the cause and to make the necessary repairs. A typical rating is 15 counts in 15 minutes. Impulse noise hits tend to be clustered with long periods between clusters; also, a typical noise impulse is much larger than the data signal, and mutilates several successive data bits. Impulse noise affects any kind of modem and any modulation scheme. It will affect more data in a high data rate stream then in a slow one because of the disruptive effect of a larger hit on the modem. The terminal equipment user should be prepared to cope with various patterns of impulse noise by providing data streams that can be acknowledged within the average interval of noise hits. If data blocks are too long, the probability is high that each one will receive a noise hit and require retransmission (with the chance of another hit. On the other hand, if blocks are short, the overhead incurred by the synchronizing sequence, header, and acknowledgments will reduce transmission efficiency.

5) Thermal Noise

Thermal noise (also called broadband or white noise) arises from the thermal agitation of electrons in resistors and in semiconductors. The greater the absolute temperature of the noise source, the greater the noise level. In telephone systems, this type of noise appears as a background hiss. Its strong appearance usually indicates a faulty component. The usual effect of thermal noise is to cause infrequent and random single-bit or double-bit errors, until a modem signal-to-noise threshold is reached, after which the error rate becomes catastrophic.

6) Intermodulation Distortion

Nonlinear components in the network will produce distortion of the data signal. The distortion becomes harmful if the time constant of the nonlinearity is short compared to the frequencies that compose the data signal. Intermodulation results in the production of spurious frequencies by a nonlinear medium. These frequencies are the sum and difference of fundamental frequencies present in the undistorted analog data signal, as well as the sum and difference between their harmonic frequencies, and between fundamentals and harmonics. The level of intermodulation distortion is typically less than 5% of the desired signal level and appears as correlated background noise that can cause characteristic distortion of the data signal (i.e., weak bit patterns).

7) Delay Distortion

In the process of transmission, all frequencies may not arrive at their destinations with the same relative phase relationships that existed at the transmission point. The effect is that some portions of the data signal appear to be delayed in time, with reference to other portions of the data signal. A modulated signal has a fundamental frequency plus harmonic frequencies. Each harmonic wave has a fixed phase relationship with the fundamental wave that must be preserved if the data signals are to be reproduced by the receiver. When the harmonics are advanced or retarded in phase, they appear to arrive early or late. Some signal components overlap the time domains of adjacent signal elements, causing intersymbol interference. The solution to the delay distortion problem is called delay distortion.

8) Incidental FM

This phenomenon is peculiar to frequency-based carrier equipment, for example, in a link with analog microwave equipment. Incidental FM usually arises from the influence of varying power supply voltages on the oscillators used in the equipment. If the oscillators are not stabilized, hum frequencies and lower frequency ringing voltages can modulate the oscillators, both in frequency (or phase) and in amplitude. The amplitude effect is usually of lesser consequence. The effect of phase jitter on a data signal is to decrease the certainty of detection of data bits in the receiving modem. High-speed modems that use PSK are most vulnerable. The pool of analog carrier systems is becoming smaller in the United States, as fiber is increasingly deployed, diminishing the importance of this impairment.

9) Frequency Error

This phenomenon, like incidental FM, is endogenous to carrier frequency equipment. It results from the fact that the heterodyning oscillators in location A can differ in the frequency by several hertz from those in location B, with the result that the data carrier and wideband frequencies are offset. This is not noticeable to a telephone listener, but modem performance can be degraded slightly, and a poorly designed modem will fail to receive data correctly.

(I) Line Conditioning

Tow option are available to reduce the effects of attenuation and delay distortion in a traditional data circuit: conditioning and equalization. Line conditioning may be used when data transmission on dedicated voice-grade lines occurs at speeds of more than 4800 b/s. line conditioning will ensure that the circuit conforms to transmission specifications and is less susceptible to errors. Line conditioning is provided on leased lines by carriers at a monthly fee. The carrier may add special equipment to the circuit; conditioning provides a method to diminish the problems of attenuation and delay, but it dose not totally eliminate the impairments; it provides for more consistency across the bandwidth. For attenuation, the carrier adds equipment that attenuates the frequencies in the signal that tend to remain at a higher level than others. Thus, attenuation still occurs but is more evenly distributed across the channel. Conditioning is not available over the switched telephone network.

Two types of line conditioning are available in the United States: C-conditioning and D-conditioning. C-conditioning is used to deal with attenuation distortion, which occurs when the relative amplitudes of the different frequency components change during transmission. This can occur because of uneven attenuation or uneven amplification. D-conditioning is used to deal with harmonic distortion, which is caused by the presence of unwanted harmonics from the input signal in the output signal. D-conditioning also improves the S/N. the standard specification requires a S/N of not less than 24 dB, a second harmonic distortion of not more than 25 dB, and a third harmonic distortion of not more than 30 dB. With D-conditioning, the carrier gives a S/N ration specification of 28 dB, a signal to second harmonic distortion ration of 35 dB, and a signal to third harmonic distortion of 40 dB. D-conditioning is accomplished by eliminating or avoiding noisy facilities for the given circuit.

An attenuation equalizer adds loss to the power frequencies of the signal because these frequencies decay less than the higher frequencies in the band. The signal loss is then consistent throughout the transmitted signal. After equalization is applied, amplifiers restore the signal back to its original level. A delay equalizer compensates for total signal delay. The higher frequencies thus may reach the receiver ahead of the lower frequencies. Consequently, the equalizer introduces more delay to these frequencies to make the entire signal propagate into the receiver at the same time.

With the recent development of sophisticated modems able to cope with problems on dialup lines (and thus, by extension, in private lines), and with the continued introduction of digital lines, the importance of conditioning, as currently defined, may diminish in the 1990s.

(m) Data Error Detection and Correction

Data communication systems require error control mechanisms to deal more explicitly with the problems listed in the previous section than with line conditioning. In a metropolitan environment with considerable man-made electromagnetic noise and multipath problems, the need for error correction is imperative, especially for data carried on copper loops or radio systems.

The original work on error-control coding was undertaken in the 1940s and 1950s by Shannon, Hamming, and Golay. Coding is now a mature branch of communication, with a strong mathematical foundation. These advances in theory combined with the emergence of inexpensive VLSI provide cost-effective means to achieve efficient and highly reliable communication.

In terrestrial links (where the signal propagation delay is small), the commonly used method of error control is the automatic repeat request (ARQ). Once an error is detected, the receiving system asks for retransmission from the sender (automatic refers to the fact that no user intervention occurs). Two types of ARQ systems exist: (a) stop-and-wait ARQ and (b) continuous ARQ. In the stop-and-wait ARQ, the sender system waits for an acknowledgment from the receiver system on the status of the transmitted block. If the acknowledgment is positive, than the sender system transmits the next block; if it is negative, it repeats the transmission of the blocks that have errors. This approach unsuitable in a satellite environment where a round-trip delay of half a second is required for the reception of the acknowledgment. A more acceptable form of ARQ is the continuous ARQ. With this approach the sender sends the blocks and receives the acknowledgment continuously. Once a negative acknowledgment is received, the transmitter sends either the block with an error and all blocks that follow it, or sends only the block that has the error.

Both of these ARQ approaches require a duplex channel, so that handshake information can be sent back and forth. If the channel is simplex, as is the case with the FM-sub carrier (or the propagation time is so long that the channel is effectively simplex-for example, communicating with an interplanetary spaceship), then a totally different method is required. The form of error control commonly used in satellite systems is the forward error correction (FEC). In this system, extra bits of data are added to the blocks for error checking and correction. If an error is detected, enough redundant information is carried along, which permits the receiving end to fix the incoming message; the receiver need not go back to the sending party to obtain a retransmission as in ARQ.

In FEC, one wants to be able to remove all redundancy from the source of the information, so that the amount of data to be transmitted is minimized. The channel encoder performs all the digital operations needed to prepare the source data for modulation. The encoder accepts information at rate R_s, and adds its own redundancy, producing an encoded data stream at a higher rate R_c. There are two types of FEC methods, the block codes and the convolutional codes.

When using a block code, the encoder accepts information in sequential k bit blocks and, for each k bit, generates a block of n bits, with $n \geq k$; the n-bit block is called a code block or code word. The ratio k/n is called the rate of the code. Thus, the stream of data is broken into k symbols of information and n – k redundant symbols for error control, where n is the total length of the code word. The resultant system is referred to as (n,k) block code, of which many forms exist; the most popular ones are the cyclic codes, which can simply be formed from cyclic shifting of bits of data in a block, thus creating a code word. The two most common coding techniques are the BCH (Bose-Chaudhuri-Hocquenghem) code and the Golay code.

Convolutional codes are another class of FEC methods. For encoding with a convolutional code, the encoder accepts information bits as a continuous stream and generates a continuous stream at a higher rate. The information stream is fed to the encoder b bits at the time (b typically ranges from 1 to 6). The encoder operates on the current b-bits and some number k (called constraint length) of immediately preceding b-bits inputs to produce B output bits, with B > b. Here the code rate is b/B. the encoder for the convolutional code might be thought as a form of digital filter with memory extending k – 1 symbols in the past. A typical binary convolutional code will have b = 1, B = 2 or 3, k = 4 or 5 or 6 or 7 (in special situations k can be as high as 70). The channel decoder undertakes the conversion of the demodulator output into symbol decisions, which reproduce as accurately as possible the data that were encoded by the channel encoder. The most widely used convolutional code is the Viterbi code. Viterbi coders that transmit 256 kb/s are now available for less than $90 in quantities.

Trellis code modulation (TCM) is a relatively new technique now available in high-speed modem. The method is such that the signal (derived and coded from the user data bit stream) is allowed to assume only certain characteristics (states). User bits are interpreted such that only certain of the states are allowed to exist from prior states. The transmitting device accepts a series of user bits and develops additional (restricted) bit patterns from these bits. Moreover, a previous user bit pattern (a state) is used to determine the current bit patterns (states). Certain other states are not allowed and are never transmitted. The transmitter and receiver are programmed to understand allowable states and the permissible states transitions. If the receiver states and state transitions differ from redefined conventions, we assume that en error has occurred in the circuit. By convention, the transmitter and the receiver know the transmission states and permissible state transitions; the receiver analyzes the receiver signal and makes a "best guess" as to what state signal should assume. It analyzes current states, compares them to previous states, and makes decision as to the most relevant state. If effect, the receiver uses a path history to reconstruct damaged bits. Trellis coding is an error-correction code with a memory. It increases the BER performance on a line by two to three orders of magnitude.

(n) Parity Checking

Parity checking is a simple but relatively unreliable method for basic error detection. It was primarily used in the late 1960s and early 1970s. Parity check schemes are simple examples of block codes. Here the encoder accepts k information bits and appends a set of r parity check bits, derived from the information bits according to sum predefined algorithm. The information and parity bits are transmitted as a block of n = k + r bits. A typical parity code is (8,7). Single parity check codes lack sufficient power to provide reliable communication. Hamming codes provides more powerful block codes and are also used for error control in computer memorize and other mass storage systems.

Two versions exist: an odd-parity and an even-parity. In even-parity, the number 1s in the 7-bit ASCII representation of each of each character being sent is counted. If that number is even, a 0-bit is concatenated to the 7-bit code; if the number is odd, a 1-bit is concatenated to the 7-bit code, thus giving an even number of bits (odd-parity does the reveres). For example, the code 1111111 becomes 11111111. The receiving end will count the number of ones; if there are an odd number of ones, the end system concludes that an error must have occurred in transmission.

The problem with this method is that line hits and dropouts tend to occur in bursts, affecting several contiguous bits, particularly at high speed. If more than one bit is affected, the method may fail to detect an error. For example, 11111111 becomes 00000011, the receiving end will not be able to detect the problem.

(o) Cyclic Redundancy Checking

Cyclic redundancy checking is the prevailing method used in conjunction with ARQ to detect errors in long blocks of data. The process of generating a CRC for a message involves dividing the message bye a polynomial, producing a quotient and a remainder. The remainder, which usually is two characters (16 bits) in length, is appended to the message and transmitted. The receiver performs the same operation on the received messages and compares its calculated remainder. If the two CRCs fail to the match, the protocol causes the block to be to be discarded, and a retransmission is requested. In contrast with the parity method discussed above, and other methods used in the 1970s (vertical and horizontal error checking), the undetectable error rate for CRC-protected data is extremely small (the undetectable error rate depends on the length of the CRC and the length of the data block).

2.3.2.3 Switching

Switching is as important for data communication as it is for voice communication. Six types of switching applicable (but not exclusively) to data are described below.

(a) Circuit Switching

In a circuit switched connection, the end-to-end path of a fixed bandwidth exists only for the duration of the session. The destination is identified by an address; the network receives the address from the sender and sets up a path, typically within seconds, to the destination. At the completion of the call, the path is taken down. Circuit switching is not only suitable for voice transmission, which employs this method almost exclusively, but also for data transmission.

Circuit switched connections are economical if the end-to-end session is short (on the order of minutes), or the geographic area to be covered is wide. However, if the sessions are long and traffic to a specific destination is heavy, the cost of using circuit switching may exceed the cost incurred by other methods (for example, a dedicated line). If calls are very short, the setup and teardown overheads may be excessive; other methods (such as packet switching discussed below) may be better.

(b) Channel Switching

Channel switching refers to a service that allows the user to establish a channel that can stay in place for hours or days, and can be reterminated (typically in minutes) as needed. The nomenclature used to describe this service varies, and the terms "reserved," "semipermanent," and "permanent" have all been used. Channel switching can be considered an extension of circuit switching with the following four variations: (1) the call setup time is in minutes rather than in seconds; (2) the duration of the call is in hours or days rather than in minutes; (3) it employs "slow" switches in the CO, such as a digital cross-connect system, rather than a traditional circuit switch; and (4) it is cheaper for the user, compared to straight circuit switching, for long session. Channel switching may be provided under ISDN, although manifestations appeared in the early 1980s using digital cross-connection systems. It is a service positioned between circuit switching and dedicated lines (no switching).

(c) No Switching

Many data applications use dedicated lines (also known as a "private line") that do not include carrier-provided switching. The line is leased at a fixed monthly rate, and the charge is independent of usage. Once installed, these facilities can stay in place for years. T1 circuits now commonly used as backbone data networks, are examples of dedicated channels. Systems Network Architecture (SNA), predominantly employs dedicated lines, although a packet switched interface is also available.

(d) Packet Switching

Early approaches to data communication were based on techniques for voice communication such as circuit switching. One soon discovered that the dynamic allocation of bandwidth would allow more efficient utilization of available network resources for interactive data communication. Packet switching has emerged as an important approach in data network. In packet switching, information is exchanged as block of limited size or packets. At the source, long messages are reassembled at the destination to reconstitute the original message. Many users can share network resources, although efficient use of transmission resources increases the network complexity. Data buffers are needed at each node; however, the storing is typically transient in nature, and should be of the order of tens or hundreds of milliseconds. Packet switching can be viewed as a case of message switching, except that the very formal procedures are used (absent in traditional message switching), and the period of nodal storage is small, as indicated.

Packet switching comes in two types: connection-oriented (such as in traditional x.25 wide-area networks), and connectionless (such as in LANs and MANs).

(e) Message Switching

Message switching refers to a method of storing a message at intermediary nodes in the network for nontrivial amounts of time (i.e., more than a couple of minutes). This method was commonly used in telegraphy and telex networks. It almost disappeared, but may now be reemerging in store-and-forward E-mail, particularly with the message handling system.

(f) Distributed Switching

LANs, MANs, and IBM's SNA, among other systems, employ a form of a distribution "self-switching."* In this environment, the data are labeled with an address and are broadcast to every user connected to the particular transmission system at hand (for example, in a LAN this would be every user on the bus or ring). Each user's equipment receiving the message examines the addressed on it and is trusted not to display or throw away the data unless the message is addressed to that user. This technique is become prevalent, and it has the advantage of being efficient because it relies on distributed user-provided intelligence to carry out the message sorting task; the sorting and switching task would have to be done centrally, by carrier-provided resources. Additionally, it can be more reliable, if properly designed, because failures of a single component dose not necessarily affect the entire communication system.

2.3.2.4 Layered Protocols

(a) Motivation

As we discussed at the beginning of this chapter, communication involves by definition two (remote) entities, also called end systems. The tow entities should be peers, namely, enjoying the same set of communication privileges, although this is a relatively new approach (throughout the 1960s and 1970s, a master-slave approach was much more typical). To undertake communication, a fairly large number of functions must be carried out. In addition, agreements must exist between the two end systems on how to undertake functions that have remote importance. These agreements are now known as protocols, and publicly agreed protocols are referred to as standards.

A layer is a defined set of related communication functions. Protocols describe ways in which remote peers can utilize functions within a layer. Layering (or modularization) provides the following benefits, among others:

1. Easier understanding of the communication process by working with a small number of logical groupings;
2. Collecting related functions in the same groupings minimizes the number of interactions between layers and simplifies the interfaces;
3. Layers can be implemented differently and changed to take advantage of new developments without affecting the other layers;
4. Simple layer boundaries can be created with at most only two neighbors.

   (Layering by itself dose not necessarily imply peer-to-peer communication capabilities: a number of vendor-specific architectures of the 1970s employed the layering concept (i.e., IBM's SNA, DEC's DECNET); however, the open-layered architectures that have evolved in the 1980s can provide peer-to-peer capabilities.)

(b) Open Systems Interconnection Reference Method

To facilities interconnection, standards for open systems have been developed by the International Organization for Standardization. Seven major layers have been defined in what is now known as the Open Systems Interconnection reference model (OSIRM), which has been available since 1984, as follows: application (7), presentation (6), session (5), transport (4), network (3), data link (2), and physical (1) layers. See Table for a listing of some of the key functions of the OSIRM layers. (The application layer should properly have been named "application support layer" because the ultimate user application utilizing communication facilities to interact with partners in other systems resided above the application layer)> this model is described in specification ISO 7498 and also in CCITT X.200. the term upper layers are in the direction of the layer (3) through (1).

All contemporary descriptions of data communication (and telecommunication, for that matter), including network management, security, addressing, and internetworking, employ the frame work defined in the OSIRM. The higher adjacent layer is called the user; the lower one is called the provider describe, respectively, the relationship between the consumer and the producer of a layer service. As one moves through layers, users become providers and vice versa.

Table 2.4 Layers of the Open System Interconnection Reference Model

Layer	Function
Application	Support of user functions such as file transfer, transaction processing
Presentation	Transfer syntaxes (character coding)
Session	Coordination services, dialogue, synchronization
Transport	Reliable end-to-end communication
Network	Delivery within a single sub network; end-to-end, such as addressing and internetworking
Data Link	Delivery of blocks of data between two points
Physical	Bit transmission

Entities exist in each layer. An entity is an active element within a layer that carries out the layer's prescribed functions. A layer may contain multiple entities for different functions. Entities in the same layer, but in different systems, that must exchange information to achieve a common objective are called "peer" entities. Peer defines an equivalent layer entity in another system within the open environment. A peer may be of a different hardware or software environment, but behaves consistently in all cases. Entities in adjacent layers interact through their common boundary.

The application layer can be viewed as an extension of local operating system supervisory functions to another system. These supervisory functions include the following: (1) identifying the intended partner and activating authentication procedures; (2) agreeing on quality of service, security, payment, etc.; (3) supplying services to control the modification of shared data; (4) determining that required resources are available; and (5) specifying more detailed application requirements.

The upper layers – application, presentation, session, and transport – are generally, although not always, independent of the telecommunication network; the reason for the exception is that some carriers may offer functionality above the network layer, for example, E-mail. In general, however, these layers are components of the end-user systems and are insulated from networking operations. The interworking layers create computer-system-computer services operating across any combination of subnetworks. The OSIRM layering is applicable both when communicating over a carrier's network and when communicating over a private (CPE) network.

Communication with a remote peer, at the same layer, involves a protocol. Adjacent entities communicate by exchanging primitives with each other via the service access point (SAP). The SAP is a conceptual delivery point, and as such it can be addressed. Note, however, that the relationship between SAPs and entities is not one to one.

(c) (N)-Layer Formalism

Some formalism related to important concepts associated with a layer, already described above in more intuitive form. This formalism will be required when discussing name and addressing issues. The functionality of a generic (N)-layer is provided by one or more (N)-entities. These (N)-entities must be identified and located to perform communication. An (N)-title is a name that uniquely identifies a particular (N)-entity throughout the (N)-layer in an OSI environment; an (N)-title is independent of the location of an (N)-entity. (N)-service access points represent the logical interfaces between (N)-entities and (N+1)-entities. An (N)-SAP address is a name that identifies a set of one or more (N)-SAPs to which an (N+1)-entity is attached. An (N)-SAP address relates to an (N)-SAP, and not directly to an (N+1)-entity. Hence, an (N)-entity is identified by an (N)-title, independently of the (one or more) (N – 1)-SAPs to which the (N)-entity may be bound. An (N)-entity is located by specifying the (N – 1)-SAP address of the (N – 1)-SAPs to which the (N)-entity is bound. A mechanism exists at each layer which associates (N)-entities with their (N – 1)-SAPs; this is the (N)-directory. The (N)-directory maps (N)-title of (N)-entities onto the (N – 1)-SAP address through which they communicate. An OSI address comprises nested (N)-SAP addresses.

(d) Primitives and Services

A primitive is the smallest unit of action that can be specified. More precisely, it is a conceptual instruction in a layer service. The primitives represent, in an abstract way, the logical exchange of information between a layer and the adjacent layers; they do not specify or constrain implementation. Examples of primitives are: establishing communication with a remote peer, sending data, and inserting a synchronization point. Most communication activities require a number of primitives to complete their tasks. A service primitive consists of a name and one or more parameters passed in the direction of the service primitive. The name of the service primitive contains three elements: (1) a type indicating the direction of the service primitive, (2) a name, which specifies the action to be performed, and (3) an initial (or initials), which specifies the layer (or sublayer) providing the service.

Two kinds of the services are available: confirmed and unconfirmed. A confirmed service produces information from the remote peer entity on the outcome of the service request (this may be needed when additional action is contingent on a successful outcome). An unconfirmed service only passes a request along; this is a faster interaction because no overhead is involved with the response.

Four generic types of service primitives for confirmed service are:

1. Request: a service request from a higher layer to a lower layer (more formally: a primitive issued by a service user to invoke a service element);

2. Indication: a notification from a lower layer to a higher layer that a significant event has occurred (more formally: a primitive issued by a service provider to advise that a service element has been invoked by the service user at the peer service access point or by the service provider);
3. Response: the response to a request (more formally: a primitive issued by the service user to complete at a particular service access point, some service element whose invocation has been previously indicated at that service access point);
4. Confirm: message passed from a lower layer to a higher layer to indicate the results of a previous service request (more formally: a primitive issued by a service provider to complete, at a particular service access point, some service element previously invoked by a request at that service access point).

Peers exchange protocol data units (PDUs) containing (1) protocol control information (PCI) and (2) data. A user initiates activity by issuing a service request across the SAP. The entity receives request and constructs a PDU, the type and values of which are determined by the request and locally available information. The PDU is delivered to the remote peer partner using the services of the underlying layers (the PDU will be enclosed as data in a subsequent service request to a lower layer. When the remote entity receives the PDU, it generates a primitive that it passes upward via the SAP to the user.

More details on the OSIRM and related standards are provided; however, a basic understanding is required at this juncture for use in the intervening chapters. ISDN, broadband ISDN, signaling, LANs, MANs, network management, and security all require an understanding of the OSIRM and related recommendation. We describe the first three layers of the OSIRM below. The entire model is described.

(e) Layer 1 – Physical

This layer deals with the physical connection of the circuits. Recommendations at this layer standardize pin connections between DTEs and DCEs and are also concerned with the transmission of bits between machines. Examples of protocol standards for this layer are RS-232-C and RS-449, noted earlier for connection between terminal devices and modems, and I.430 for ISDN.

(f) Layer 2 – Data Link

This layer deals with data transmission over a signal link. The protocol in this layer detects and corrects in the transmission. This layer creates frames for the data to be transmitted, including a CRC code. The functionality includes sending acknowledgments to the sender end system. It will signal for retransmission if a frame is out of sequence or is mutilated. This layer uses flags and headers in the frame, so that the receiver and system can recognize the start and end of a frame. Typical standards for this layer are: IBMs Synchronous Data Link Control (SDLC), ISO's High-level Data Link Control (HDLC), and ISO's Link Access Procedure B (LAP-B).

(g) Layer 3 – Network

This layer receives message from the higher layer, segments them into packets, and sends them to the receiver through the data link physical layer. At the receiving end, functionality in this layer reassembles the message into original form. It enables routing, multiplexing, and flow control. Typical standards for this layer are: CCITT's X.25 Packet Level Protocol and ISO's 8473, which deals with internetworking.

2.4 Telecommunication Carriers

This section describes the commercial providers of telecommunication services. In the United States, a common carrier is an agency that provides telecommunication services to the general public. A common carrier typically provides a spectrum of services. Descriptions of services and rates are called tariffs, and are filed with the Federal Communications Commission (FCC) or with the local state public utility commissions (PUCs). Common carriers can be grouped as follows:

1. Exchange-access carriers (intra-LATA), such as the Bell Operating Companies (BOCs) and independent telephone companies; these are collectively known as LECs.
2. Interexchange carriers (IXCs), such as AT&T, US Sprint, and MCI; and

3. Specialized common carriers (SCCs), such as Telenet; SCCs are like other common carriers, but specialize in providing specific services (for example, data or video communication); these are also called value-added carriers, or value-added networks (VANs).

While 1500 or so local telephone companies provide exchange access common carrier services, 85% of the business is provided by the BOCs. Approximately 5% of the LEC's customers account for 50% of the revenues. Approximately 200 IXCs exist in the United States.

Other types of carriers also exist. A private carrier offers point-to-point links, has no switched service, negotiates with customers on an individual basis, and has a majority of leases that are long-term, with a limited and stable customer base. A company that offers only nonswitched point-to-point service to business customers, and dose not offer any services to the general public, can be considered a private carrier. These carriers are also referred to as "noncommon carriers." A nondominant fiber optic carrier is a carrier that provides principally or solely end-to-end fiber-based communication links to another carrier or to end-users, generally in an inter-LATA arrangement. Teleport is, formally, an access facility to a satellite or other long-haul telecommunication medium that incorporates a distribution network (usually fiber-based) serving the greater or other economic development. In practice, however, most teleports are not usually involved in major real estate or local economic development. About 20 teleports operate or plan to operate in the United States.

2.5 Network Design Philosophies

In reading this material, keep in mind that, depending on the objective, one may be led to design and use different network topologies, technologies, and architectures. For example, some objectives could be to:

- Minimize cost (build the cheapest network);
- Maximize the cost-performance ratio (get the most network for the dollar);
- Maximize profit (build a network that allows the firm to be very aggressive, reaching new markets, et cetera);
- Maximize profit rate (especially for a utility);

- Minimize risk of loss (military network);
- Maximize safety (network designed for police or fire departments);
- Maximize quality of services (for a given investment);
- Maximize growth opportunity for the firm (build a network that can easily grow in the future – this may be the course of action of a carrier);
- Maximize prestige of the firm (i.e., buy the newest equipment to impress investors, competitors, public);
- Many other possible objectives.

Clearly, these criteria are not all compatible with each other: a network built to minimize the cost will probably not maximize reliability and quality of service. Hence, the telecommunication manager must bee familiar with the dynamics of the design synthesis. Complexity in design (and corresponding responsibility for the planners) continues to increase inexorably. Using 1985 as an arbitrary reference for discussion and ignoring the effects of divestiture, the planner must now deal with the following new technical issues and options (some of which were only embryonic at that time):

- Availability of high-speed dialup modems operating up to 38.4 kb/s, and cellular telephony modems operating up to 16.8 kb/s, which make the optimization process compared to dedicated lines and ISDN an interesting one;
- Proliferation of personal computers and workstations in the office and on the factory floor, and their need to communicate, probably through internet worked LANs. High-speed backbone networks operating at 16 and 100 Mb/s are also becoming available. In addition to coaxial-based solutions, the planner can choose unshielded twisted-pair and multimode or single-mode fiber;
- Fourth-generation PBXs that not only allow integration of data, but also include LANs, resource servers (file servers, communication servers, protocol gateways, etc.), ISDN access, and advanced internal algorithms such as "dynamic bandwidth allocation." Also, we have witnessed the substantial improvement of CENTREX services, making it more

competitive with PBXs, and rendering the decision-making process in choosing a solution more difficult.

- Availability (in the United States) of high-speed digital interfaces to the telephone plant, to another networked PBX, or to a mainframe computer via a CPE T1 mutiplexer. T1 multiplexers are being introduced in private networks at an ever-increasing for the purpose of corporate-wide voice, data, and video integration. Fractional T1service is also becoming important, and there is talk of fractional T3;
- The introduction of high-speed channel-to-channel communication among mainframe computers, and between mainframe peripherals;
- The ongoing introduction of ISDN with its user-to-user and user-to-network signaling capability, and advanced services, including network-based automatic call distributor (ACD), and automatic number identification (ANI), which facilitates integrated voice and data applications. Switched Multi-megabit Data Service (SMDS), with data rates in the 1.5 to 45 Mb/s range, and broadband ISDN (BISDN), with data rate in the 150 to 600 Mb/s range, are also on the horizon;
- The possibility of interconnecting dispersed LANs via MANs and potentially and potentially using fiber optics facilities;
- The emergence of very small aperture terminal (VSAT) satellite networks, allowing cost-effective two-way communication over four-foot dishes, and other radio services for data usage;
- The increased need for network management: networks can cost millions of dollars every year; optimal usage is critical;
- Security has become a major area of concern as more and more computers are connected to networks, making them accessible and susceptible to attacks.

All of these factors and technologies complicate the design process; understanding the new design requirements and acquiring the appropriate tools is a mandatory effort in establishing cost-effective, reliable, and flexible cooperate networks. The responsibilities of a data communication manager are becoming more demanding as industry competition increases along with the ensuing multiplicity of available communication options and the increased obsolescence rate of this technology. In designing a network, multiple options must be analyzed and the issue of hidden costs associated with these options must be taken into account. This richness of communication options offers the data communication manager major opportunities, not only in terms of optimal network design, but also in terms of network management, grade of service, and reliability. However, analytical methods are required to facilitate the decision-making.

The multiplicity of options makes the task more difficult in the decade of the 1990s. Additionally, deployment of the least expensive network may not always be the best course of action if the network constricts the company's ability to be competitive, deliver products and services quickly, grow smoothly. As we can see, the problem is not a trivial one. Due to the high multidimensionality of the solution space.

CHAPTER 3

FUTURE DEVELOPMENTS IN TELECOMMUNICATIONS

3.1 Introduction

The 1980s and 1990s was marked by rapid development of telecommunications services and technologies. We currently use services daily that were not available ten to fifteen years ago, such as LANs, cellular phones, and graphical Internet. It is expected that this technological development and the growth of telecommunication business will continue for several years. Examples of such expected phenomena as well as the technologies and services under development that will be put into use during the coming few years are:

- High penetration of low-cost mobile services, PCS;
- Competition in local networks through the WLL technologies;
- Digital high-quality and high-capacity broadcasting systems;
- Introduction of new services using the intelligent network concept;
- Extremely high-capacity optical networks (terabits/s);
- Multimedia communications with improved quality;
- Customized media presentation;
- Interactive video services;
- Safe and user-friendly electronic shopping services;
- Speech recognition and synthesis systems;
- Pocket size "smart" mobile terminals;
- Real-time language translation;
- Video-on-Demand services;
- Electronic libraries;
- Integration of telephone service and Internet.

It is difficult to estimate which new services will get market acceptance and which will not. A technology must be available; but, in addition, success depends on many other things such as how the new services are launched and charged, what alternative services are available, and timing of the launch. In the following sections we will look at some new technologies and services in more detail.

3.2 Optical Fiber Systems

The major transmission networks will use optical fiber transmission systems that provide much higher capacity than the systems of today. The optical fibers will one day be used in the LANs just as they re used in trunk networks today.

WDM provides a higher capacity than fiber cables. Through WDM, many systems can use the same fiber if they operate at a different wavelength. At the receiving end optical signals are separated from other signals with optical fibers.

In local networks the technology known as passive optical networks (PON) may be put into use. In this technology, optical couplers are used to split and combine optical signals to and from subscribers, which is assumed to make the use of optical transmission in local networks economically feasible.

Coherent optical systems will increase the capacity of fibers dramatically. Present optical systems merely send light pulses and use the whole bandwidth of the fiber for a single signal. Coherent technology means that we use light as a carrier in the same way as we use radio frequencies in present radio systems. Then we could modulate multiple carriers to the fiber and use the whole bandwidth of the fiber efficiently.

3.3 Mobile Communications

The capacity of the cellular mobile systems will be increased with sophisticated speech processing technology, microcell and picocell structure of the networks, and new frequency bands that are put into use. Multimode mobile terminals that are able to access different networks are becoming available. These terminals may operate like a cordless telephone in an office environment, cellular telephones at different frequencies (GSM, DCS1800, PCS1900, and CDMA), and even as satellite mobile telephones.

An integrated system known as International Mobile Communications 2000 (IMT2000) or Universal Mobile Telecommunication Service (UMTS) will provide a wide range of telecommunication services to mobile subscribers. Among them are mobile telephone, data communications, facsimile, video telephone, short messaging, multimedia, and location identification. IMT2000 will use many different access technologies such as satellite communication, cellular radio, or cordless telephone technology. Services will appear the same no matter which access technology is in use.

3.3.1 Wireless Office

Wireless LANs use spread spectrum technology, TDMA radio interface (e.g., DECT technology), or infrared technology. Most of the present systems are proprietary technologies. However, there are standards such as DECT and PACS that may support the development from wireline telephones toward a wireless office. With cordless PABXs utilizing digital cordless technology the possibility of a mobile office is not far away. However, decreasing charge for public cellular service may make cellular technology more attractive than the cordless technology.

3.4 Local Access Network

Some years ago it was assumed that copper cable was a transmission media of the past and that it would soon be replaced by optical fiber. However, copper cable has a very important advantage. Almost all offices and homes are already connected to the telecommunications network with copper cable pairs. The telecommunication network I not reasonable if there are technologies that make high data rate transmission over pair cables feasible.

The recent development of transmission technology has given remarkable results, and now we can see that the telecommunications network operates may make all future services of today available via the existing pair cable network. These new technologies are together called digital subscriber line (DSL) technology.

3.4.1 High Date-Rate Digital Transmission over Existing Subscriber Pairs

As we know, there are many emerging technologies (xDSL) that increase the capacity of exiting copper cables. These DSL technologies provide simultaneous high data-rate access to subscribers in addition to ordinary telephone service. In the United States the implementation of these technologies is expected to proceed more rapidly than in Europe and they will play a key role in providing the new services, especially information services of the Internet, which are discussed in section 3.6.

The xDSL technologies will also be used in the business environment for LAN interconnections in a region and instead of conventional 1.5 or 2-Mbit/s digital systems in the telecommunications network.

3.4.2 Fiber in the Local Loop

PONs are already used in various countries. PON technology uses passive optical couplers for signal distribution to subscribers. It is expected to provide low-cost optical connections to customers' premises.

Cable-TV operators are also installing optical cables in the local access network to provide telecommunication services in addition to cable-TV. These network use SDH transmission rings with add/drop multiplexers that efficiently "add and drop" subscriber channels of a requested data rate from a STM-1 ring. These new access networks make cable-TV operators important players as suppliers of telecommunications services in the future.

3.5 Wireless Local Loop

Both cellular mobile technologies and cordless technologies are used to offer a quick and low-cost way to build up fixed subscriber connections in the area of another telecommunication network operator. WLL or RLL technologies make competition in local access networks possible, and ordinary subscribers may select the best available service provider.

3.6 Interactive Digital Services

New services will be provided to bring information or entertainment to the home and business communities. Some of the potential services are explained briefly here.

3.6.1 Video-on-Demand

The local access network can be exploited by new DSL technologies to transmit a high enough data rate for video service. Distribution quality video can already be compressed to 1.5 to 2 Mbit/s and transmitted (together with the telephone signals) over the ordinary subscriber line.

Only one video channel is required in VoD because subscribers use the service as if it were an ordinary video recorder with huge automatic tape storage. They may elect any movie from the graphical menu. The film is then transmitted to his receiver from the video server out in the network. Users have the same video recorder buttons available such as play, fast forward, rewind, or still picture and may change the film at any time.

The success of this service is highly dependent on the charge that a subscriber has to pay for the service.

3.6.2 Information Services

Internet presently provides a worldwide information service. These services will be further developed and the usage of them will become more comfortable and more affordable because of ISDN subscriber connections, DSL technologies available to subscribers, and ATM and high-rate optical systems in the trunk connections inside the network.

3.6.3 Home Shopping

Home shopping, for which ordinary mail is presently used, will use electronic catalogues in the future. This has an advantage over mail if the products can be seen in action and if question can be asked about the product. An integrated telephone would allow a customer to clarify the details of an order at the same time. Even a virtual visit to a vacation site, for example, will become practical. A customer may have a virtual walk in various hotels and select the one she likes the best.

3.6.4 Video Conferencing and Video Telephony

With the cost of traveling and the time required, video conferencing will become more popular. A special studio and semipermanent high-rate data links are no longer required for a video conferencing service. Just an ordinary PC, a plug-in video coder, and an ISDN-card together with an ISDN or other higher data rate subscriber line make video conferencing available to anybody.

3.7 Internet and Intranet

The use of the Internet as a messaging and information network expand rapidly in the 1990s. We reviewed home shopping in section 3.6 as one growing application of the Internet. The residential use of the Internet will increase as new easy-to-use access terminals and high-speed access technologies such as cable modems and DSL-technologies become widely available.

The business use of the Internet requires robust security. The usage of the public Internet network to make up secure VPNs or Intranets will increase. As security features of the Internet improve its commercial use increases further when selected Internet users are allowed to access dedicated intranets. These so-called extranets are attractive communication networks for all data interaction between a corporation, its suppliers, and its customers.

The Internet is based on an efficient packet-switched technology and has been developed to provide an acceptable grade of service for voice and video applications. We expect that the Internet will take a share of voice communication carried presently by the circuit-switched telephone and ISDN networks.

3.8 Computer Telephone Integration

A use of a computer in the place of a telephone to get access to telephone service is known as computer telephone integration (CTI). Computer software will make the man-machine interface of telecommunications network user-friendly, which is not the case today. Video conferencing is one example of the new services that are available for ordinary subscribers of ISDN service. This requires a computer equipped with an ISDN card and a video encoder. Other services supported by CTI could be integrated telephone catalogue, language translation, sending additional information (e.g., a document that calling parties are designing together), video games, and access to information services corresponding to the present Internet. With the help of CTI the present information services will be enhanced to provide an integrated telephone service that allows a user, for example, to ask additional information about the product he/she is ordering.

One very important application of CTI is the use of Internet for voice communication. This is an attractive alternative because international Internet calls are much cheaper than ordinary telephone calls.

There are many emerging technologies for the residential access to Internet such as xDSL and cable modems over cable-TV networks. When a subscriber gets a new high-speed access to the Internet, he/she at the same time has a competing alternative for voice communication. This service is very promising; but speech communication requires further development of the Internet and speech encoders.

3.9 Voice Over IP

The vast majority of information exchanged over the public telecommunication networks has been voice. The present voice telecommunication networks, public telephone and ISDN networks, use digital technology and the circuit switching principle. The circuit switching provides good quality service and does not require a complicated encoding algorithm. A simple waveform coding scheme such as PCM is sufficient for a circuit-switched connection that provides a CBR service. Charging for the voice service has been straightforward, that is, we pay for the duration of a call. This is relevant because each call reserves a certain data capacity whether there is speech on the line or not.

The characteristics of data transmission are different from waveform-coded speech, and the data networks that were developed to provide data services utilize packet-switched technology. All modern technologies for data communication such as LANs, Internet, Frame Relay, and ATM use packet-switched technology instead of the circuit-switched principle. Packet-switched networks utilize network resources more efficiently than circuit-switched networks because the capacity in the network is dynamically shared between all users. If there is no data to be transmitted between two users, their share of the data capacity is available for other users. This difference in the operation principle makes a packet-switched network superior to a circuit-switched networks.

As the importance of data communications has increased, the new technologies such as ATM are designed to support the CBR service required by traditional speech and video encoders. ATM provides a transmission mode for speech service, but we may expect that access to it will not be widely available in the near future although it will be used as a transmission technology in long-distance networks. Another packet-switched networks is the Internet. It has become very popular and access to it is available for every home with a telephone, a personal computer, and a modem or an ISDN network terminal. The evolving xDSL technologies provide even better performance access to the Internet. The ISPs provide access to the global Internet and charges for this service are based on the time of usage of the service or simply on a monthly fee. This allows subscribers to utilize international data communication networks at a cost level close to a local telephone call. If the Internet could provide voice service the subscribers would be able to make international calls via it instead of the telephone network.

The implementation of voice over Internet (VOIP) service is attractive for subscribers because it reduces the cost of international and long distance calls; it is also attractive for the ISPs because it would increase the usage of the Internet service. The technology for VOIP does not yet provide as good voice quality as a circuit-switched telephone network, but there is a lot of activity developing protocols and speech encoders for the implementation of the high-quality voice service. In addition to the IETF activities, there is even a VOIP Forum with more than 100 vendors to speed up this development. This consortium has approved the first standards for the voice encoding in the beginning of 1998 for the telephone calls over the Internet. One problem is that the Internet is designed for data communications and the packets suffer a long and variable delay that decreases voice quality. To overcome this problem the protocols of the Internet are being developed to provide a certain share of network resources for each voice call through the network. Figure 3.1 shows three possible ways to make a call over the Internet.

In the first application example in figure 3.1 a telephone subscriber dials the number of a local gateway and a call travels over the PSTN to the nearest gateway. Then the caller enters the destination telephone number and the call travels over the Internet to the gate way nearest the specified telephone. The routing and speech processing is performed by the local gateway. From the remote gateway, the call is then sent over the PSTN to the destination telephone. Now the Internet carries a long-distance section of the call instead of the PSTN.

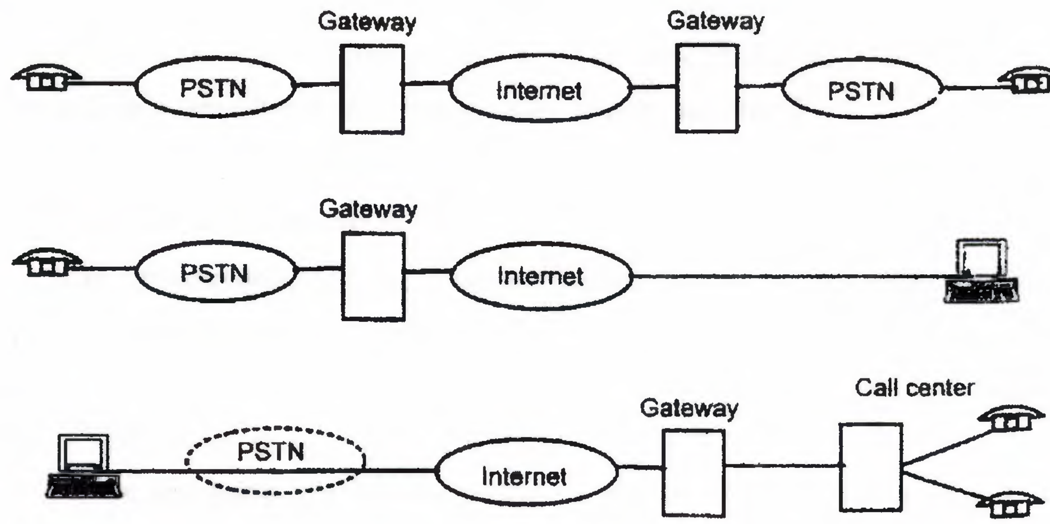


Figure 3.1 Voice over Internet application.

The second application illustrates a call established from an ordinary telephone to the remote computer connected to the Internet. If the destination computer is permanently connected to the Internet, the telephone subscriber tells the destination Internet address to the gateway using the telephone keypad. The call then travels directly to the desired computer. If the computer uses PSTN for access to the Internet, then the gateway function is needed in the destination end as well and the telephone number is used to specify the destination, just as in the first example.

The third example in Figure 3.1 illustrates an enhancement to the WWW service. People surfing the Web can connect to a company's call center by clicking "Call" button located on the company Web page. Users can communicate with a customer service group, ordering department, or help desk using their Web browser and a personal computer equipped with a compatible speech encoder. This is an important new feature as the commercial use of the Internet expands.

Many propriety technologies for these services are available and we expect that these applications will expand as standards mature. The Internet will also be widely used for facsimile calls and video conferencing as standards evolve.

Because packet-witched technology can deliver services far more cost efficiently than today's circuit-switched technology, there are standards under development for voice services over other data networks as well. We may expect that good quality voice service will soon be available over LANs, Frame Relay, and ATM networks.

3.10 Broadband ISDN and ATM

The asynchronous transfer mode is specified to be a transmission method for B-ISDN in the future. ATM is designed to support any type of information transfer like speech, video, or data. Present networks are primarily designed for CBR service (e.g., of ISDN) or VBR service (e.g., Internet or LANs).

At the beginning of the conversation the service type is specified (e.g., CBR or VBR). For speech or video, CBR capacity is reserved fo0r the whole duration of the connection. All types of information is packed into small fixed-sized cells. These cells are transmitted over SDH links between ATM switches via virtual circuit that is built up in the beginning of the conversation. ATM switches route them quickly according to the circuit identification in each cell.

ATM technology will first become popular in high-rate data communications. But it will take many years before it will be widely used for other services such as speech and integrated services of B-ISDN.

3.11 Digital Broadcasting Systems

Present broadcasting systems such as radio and TV use technologies that were originally developed in the 1950s. Even though some updates have been made such as color TV and stereo sound, present systems do not meet the quality requirements of today and the future. Another problem with these systems is that they not utilize radio frequencies as efficiently as more modern technologies could do.

3.11.1 Digital Radio

Digital broadcast radio will be introduced in a few years to come. Digital broadcasting technology will improve present FM-radio transmission quality to the quality level of compact disc. In will also remarkably increase the capacity of the broadcast radio band.

3.11.2 Digital TV

Digital TV will also be introduced in a few years. It will improve quality and provide some additional services, but its main advantage is the more efficient use of radio frequencies. An additional converter or a new TV set will be required. The introduction of digital TV may postpone the introduction of HDTV.

3.11.3 High-Definition TV

High-Definition TV (HDTV) will improve the quality of TV to the level of motion pictures. There has not yet been a significant enough market demand and the introduction of HDTV technology has been delayed. The costs of the technology of receivers as well as the cost of the distribution network seem to be too high for a successful market launch in the near future.

3.12 Summary

New transmission and witching technologies like SDH and ATM will allow voice, data, or video traffic to be transported easily over the same network with a variety of speeds. These technologies are the basis of the telecommunications network of the future. The usage of the Internet as an information network will expand; it currently supports services provided by the PSTN and ISDN.

The development of new technologies for local access networks makes new high-quality services available for business users as well as for ordinary telephone subscribers. Potential examples of these services are video conferencing, VoD, and the supply of high-performance information services.

The cost of telecommunication services will decrease because of the deregulation of telecommunications markets, and the usage of the services will expand. The Internet will offer an attractive alternative to PSTN. The new technologies and competition will make mobile telecommunication service available to anyone in developed countries.

All of these new telecommunication technologies will be combined to form the "Information Superhighway". Only time will tell what services will be created, which of them will become popular, and how will they change our everyday life.

CONCLUSION

Telecommunications network can be viewed as an assemble of a number of links, like LAN, MAN, WAN and the telephone network. We explained the operation of a public-switched telecommunications network (PSTN) and cited the conventional telephone networks as an example, besides that the ways in which data or voice are converted to signals to be transferred between two points via the links. These two points are connected together with links connected by nodes and switches.

Switching system or switching system allows a user to connect with any other subscriber by dialing the subscriber's number, and it eliminates the need for point-to- point wiring, for data transfer switching system provides an internal transport structure for carrying information from one node to another.

Many physical factors inhibit successful data transmission, some of the factors applicable to a traditional telephone channels are attenuation, cross talk, thermal noise, echo, etc.

Data error detection and correction for the transferred data.

To avoid the physical factors which inhibit the successful data transmission, digital networks, fiber-based networks and networks designed specifically for data communications.

Just as the fax changed the way we worked in the past, video conferencing via ISDN looks set to dominate office communications; in the future it allows computers to be time efficient with improving productivity.

In the future some new services and options will be available for the human being to use it for his comfortability like deliveries from shops for house keeping.

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